Radius Console System

Installation & User's Guide

AXIA
A TELOS COMPANY

Manual Rev 0.96 — March 2011
for Radius v1.00 and higher

p/n 1490-00070-001

USA Class A Computing Device Information To User. Warning:

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, Subpart J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense.

NOTE: Objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of only shielded cables.

Canada Warning:

"This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications." "Le present appareil numerRadiusue n'emet pas de bruits radioelectrRadiusues depassant les limites applicables aux appareils numerRadiusues (de les Class A) prescrites dans le Reglement sur le brouillage radioelectrRadiusue edicte par le ministere des Communications du Canada."

CE Conformance Information:

This device complies with the requirements of the EEC Council Directives: 93/68/EEC (CE Marking); 73/23/EEC (Safety – low voltage directive); 89/336/EEC (electromagnetic compatibility). Conformity is declared to those standards: EN50081-1, EN50082-1.

Important Safety Information

To prevent risk of electric shock: Disconnect power cord before servicing. If fuse replacement is required, please note: For continued protection against fire, replace fuse only with same type and value.

Caution

The installation and servicing instructions in the manual are for use by qualified personnel only. To avoid Electric Shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel.

Electrical Warning

To reduce the risk of electrical shock, do not expose this product to rain or moisture. Keep lRadiusuids away from the ventilation openings in the top and rear of the unit. Do not shower or bathe with the unit.

This equipment is designed to be operated from a power source that includes a third "grounding" connection in addition to the power leads. Do not defeat this safety feature. In addition to creating a potentially hazardous situation, defeating this safety ground will prevent the internal line noise filter from functioning.

Ventilation Warning

The Axia Radius Core requires the free flow of air for adequate cooling. Do not block the ventilation openings in the top and rear of the unit. Radius Core must be mounted with a blank rack space above or damage may occur.

Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

Customer Service

We support you...

By Phone/Fax.

- You may reach our 24/7 Support Team anytime around the clock by calling +1 216-622-0247. For billing questions or other non-emergency technical questions, call +1 216-241-7225 between 9:30 AM to 6:00 PM USA Eastern Time, Monday through Friday.

By E-Mail.

- Non-emergency technical support is available at **Support@AxiaAudio.com**.

Via World Wide Web.

- The Axia Web site has a variety of information which may be useful for product selection and support. The URL is **http://www.AxiaAudio.com**.

Feedback

We welcome feedback on any aspect of the Livewire products or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.

Updates

The operations of Radius and Radius Core are determined largely by software. Periodic updates may become available - to determine if this is the case, visit our web site periodically, or contact us for advice concerning whether a newer release is more suitable to your needs.

We recommend joining the Axia Tech mailing list to ensure that you receive immediate updates regarding new software or documentation releases. You can do so by clicking the "Sign Up Here" links at the top of the Axia Download and Manuals pages at AxiaAudio.com/manuals/ and AxiaAudio.com/downloads/.

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Warranty

This product is covered by a five year limited warranty, the full text of which is included in the rear section of this manual.

Service

You must contact Axia before returning any equipment for factory service. Axia will issue a Return Authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer at Axia. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers please contact Axia technical support at +1 (216) 241-7225. All other customers should contact their local representative to arrange for service.

We strongly recommend being near the unit when you call, so our Support Engineers can verify information about your configuration and the conditions under which the problem occurs. If the unit must return to Axia, we will need your serial number, located on the rear panel.

Credit Where Credit's Due

It's a pleasure to be a part of this organization. Thanks to all of the engineers, architects and designers that made this console a reality - you know who you are. *cn*

Feedback is welcome

At Axia, we love to hear your feedback. If you find anything in this manual that you feel needs clarification or correction, please let us know by sending an e-mail to cnovak@AxiaAudio.com .

About This Manual

This manual covers the details of the Radius control surface and Radius Core. Radius is designed to operate as a standalone console; however, it also has network capabilities. To learn more about these concepts, you may wish to read ***Introduction to Livewire: System Design Reference and Primer*** and ***Audio Over IP: Building Pro AoIP Systems with Livewire*** by Steve Church and Skip Pizzi, available from Elsevier Press.

In these publications we explain the ideas that motivated Livewire and how you can use and benefit from it, as well as nitty-gritty details about wiring, connectors, and the like. Since Livewire is built on standard networks, we also help you to understand general network engineering so that you have the full background for Livewire's fundamentals. After reading this reference material, you will know what's up when you are speaking with the network guys that are often hanging around radio stations these days.

New in this Manual

This manual, written in March 2011, details the workings of the Axia Radius console and Radius Core. Although we strive for accuracy, some features and operational characteristics may differ in actual use from their descriptions herein. We invite feedback and corrections from our clients.

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A Note From The CEO of Telos

In 2003, Telos launched the Axia Audio division based on an idea. That idea sparked a whole new approach to radio consoles, using Ethernet as a high-speed expressway – the routing infrastructure for audio and data inside the radio station.

Howard Aiken, the scientist responsible for the conceptual design of IBM's groundbreaking Harvard Mark I computer, once advised "Don't worry about people stealing your ideas. If your ideas are any good, you'll have to ram them down people's throats." Sure enough, during our first few years, the idea of using Ethernet to transport audio was so novel that we spent most of our time simply telling people our story, sleeping in hotel beds – and bringing home very few sales.

After what seemed like an endless stream of hotel desks and airline food the idea took hold, thanks to forward-thinking engineers who understood the power of using data networks for audio transmission. Our first installations proved that the technology was real. Then those engineers began sharing stories of their own, telling other folks how much they loved what Axia did for them. Word got around, and we sold thousands of consoles (enough that Element consoles are now one of the best-selling broadcast boards ever).

As more and more broadcasters embraced IP, dozens of companies partnered with us to make innovative broadcast products that connect to Axia networks with just the click of an Ethernet cable. And we conceived of some interesting new devices and applications to help broadcasters take full advantage of their networked environment.

For example, we introduced broadcast VoIP telephone systems that integrate with consoles, merging telephony into the studio network without adding infrastructure (and giving broadcasters the world's first consoles with integrated Telos hybrids). Intercom systems that connect via Ethernet, and permit users to take broadcast-quality intercom calls to air instantly. Networked codecs and audio processors like the Telos Zephyr iPort and Omnia.8X, which concentrate multiple audio devices into a single network node, significantly changing the economics of complex systems.

And along the way, we expanded our talented staff of engineers, scientists and broadcast professionals, growing the largest R&D organization in the industry. We're also the first and only broadcast equipment maker with a 24/7 support team that's ready to offer assistance any time of the day or night. As they say, radio never sleeps – so neither do we.

We realize that it's not due to our efforts alone that these things have happened — it's due to yours. We owe quite a lot to you, and we will never, ever forget it. You are, literally, the reason we are here! And we will continue to innovate, explore, think and invent, for one simple reason: we love to hear you say "Wow!"

Michael "Catfish" Dosch

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CAT.6, small and light

yet sound and data pour forth.

What magic is this?.

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Quick Start:

Radius console and Core

The following chapters of this manual will give you an in-depth understanding of the capabilities of your new Radius console, from installation to advanced functions. This section isn't meant to take the place of the following chapters, but it will help you get everything connected quickly, and point you to the parts of the manual you'll need to get up and running with a minimum of fuss.

Radius consists of these components:

- **Radius control surface:** 8-fader frame with master controls and meters.
- **Radius Core:** Rack-mounted CPU chassis with audio I/O and GPIO connectors.
-

Radius and Radius Core.

Making Connections

You've probably unpacked your boxes and are sitting next to a pile of Axia gear, wondering what to do first. Here's what you need to know in order to get started:

1. Packed with your Radius is a cable with 6-pin Molex™ connectors on each end. This is the power/communications cable. Connect the end of the cable without the strain relief to the port on the underside of your Radius control surface. Connect the other end of the cable to the connector labeled CONSOLE on the back of your Radius Core unit.
2. Connect audio sources (microphones, CD players, etc) to inputs of your Radius Core.
3. Connect monitor amplifiers, headphone amplifiers, etc to outputs of your Core. Be sure to set all amplifiers to their minimum volume setting.
4. Follow the instructions found in Chapter 1 to configure IP address settings for your Radius.
5. Refer to the User's Manuals packed with any other Axia Audio nodes for information on how to configure IP addresses for these units.

What's Next?

When you've completed physically connecting your Axia hardware, jump to Chapter 2 for a quick tutorial on configuring audio sources. With that done, you'll be a hop, skip, jump away from being on-air! After that, you'll likely want to do some in-depth reading about all that your new console can do. Here's a brief description of the contents of the following chapters:

- **Chapter 1: Setup and Connections** explains connections and basic Radius setup.
- **Chapter 2: Radius Inputs and Outputs** explains how to generate Source Profiles and construct backfeeds for selected sources.
- **Chapter 3: Radius Operations** takes you behind the scenes to explore operator controls and options.
- **Chapter 4:Show Profiles** illustrates how to set and

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save unique “snapshots” tailored for specific shows. These Show Profiles can be instantly recalled whenever needed.

- **Chapter 5: GPIO** tells how to associate routable logic commands with any audio source.

- **Chapter 6: Advanced Controls** discusses advanced features that are configured in Radius’ configuration web pages.

- **Chapter 7: Livewire Networking and Radius** will tell you how to connect your console to a network and reap the benefits of networked studio systems.

- **Appendices** provide additional technical specifications and a complete set of block diagrams for your reference.

Read on. Enjoy your Radius.

Stay up to date! We recommend joining the Axia Tech mailing list to ensure that you receive immediate updates regarding new software or documentation releases. You can do this by clicking the “Sign Up Here” link at the top of AxiaAudio.com/manuals/ & AxiaAudio.com/downloads/.

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Chapter One:

Setup and Connections

Introduction

If you are reading this manual, you are probably the proud owner of a new Radius console. You've read the Quickstart section, are probably already passing audio, and are hungry for more knowledge. Here we go!

The Basics

Radius was designed to get you on the air as quickly as possible with the least amount of effort. It's the perfect standalone radio console (and it even networks, too - see **Chapter 7: Livewire Networking and Radius** for details on how to do that).

This chapter covers connecting Radius to your other studio gear, such as microphones, CD players and other audio playback devices. The audio outputs of the Radius Core connect to control room monitors, preview (cue) speaker, headphones amplifiers and audio recording devices. These external audio devices may be analog or AES devices.

The GPIO (General Purpose Input Output) ports provide logic control interfaces to devices such as air lights, recorders and CD players.

The 100 Mbps Ethernet ports may be used to connect PCs running Axia's IP-Audio driver, to expand your system by adding other Livewire devices such as Axia Audio Nodes, or equipment from Axia Partners (see AxiaAudio.com/partners/ for the latest list of companies whose equipment connects directly to Axia networks).

1000 Mbps (Gigabit) Ethernet ports are used to connect to other Radius Core units or an Ethernet core switch — another topic covered in Chapter 7.

A 6-pin Molex connector is used to connect your Radius mixer with the Radius Core mixing engine. The IEC receptacle is for your A.C. mains power connection.

Assigning an IP Address

Before you can configure your Radius hardware, you'll need to assign it an IP address. Press-and-hold the **Meter** and **Clock** keys for three seconds. The Studio display now shows the options and the Studio Monitor control is used to navigate, select IP Address options and change parameter values. Consult your network administrator if you are unsure about what address to use.

Each Radius Core requires a single, fixed IP address. The Radius control surface is a controller for the Core, so they share the same IP addresses. If your Radius to be a standalone console, the default IP address will probably be fine. See Chapter 7 for more details if your Radius is to be part of a larger network.

Notes on Physical Installation

The Radius Core is fan-free, so it can be located in any studio without fear of unwanted noise. Since Radius is convection-cooled, those vents and heat sinks serve a very important purpose! Be sure to mount the Core in an equipment rack with free air flow. We recommend using a ventillated 1RU rack spacer above and below the Radius Core to ensure adequate ventilation.

To install the Radius console in your studio, all you need is a desk — no cutout required.

Radius Core: Front Panel

The indicator panel of the Radius Core, shown on the following page, is very simple. There are no controls for you to push or turn. Configuration and setup is performed from the console itself or through the Radius web browser interface.

Front Panel Indicators

Even though there are no knobs or dials, the front panel has some very useful indicators that tell you about the health and status of your Radius Core. In all cases, solid GREEN indicates normal operating conditions.

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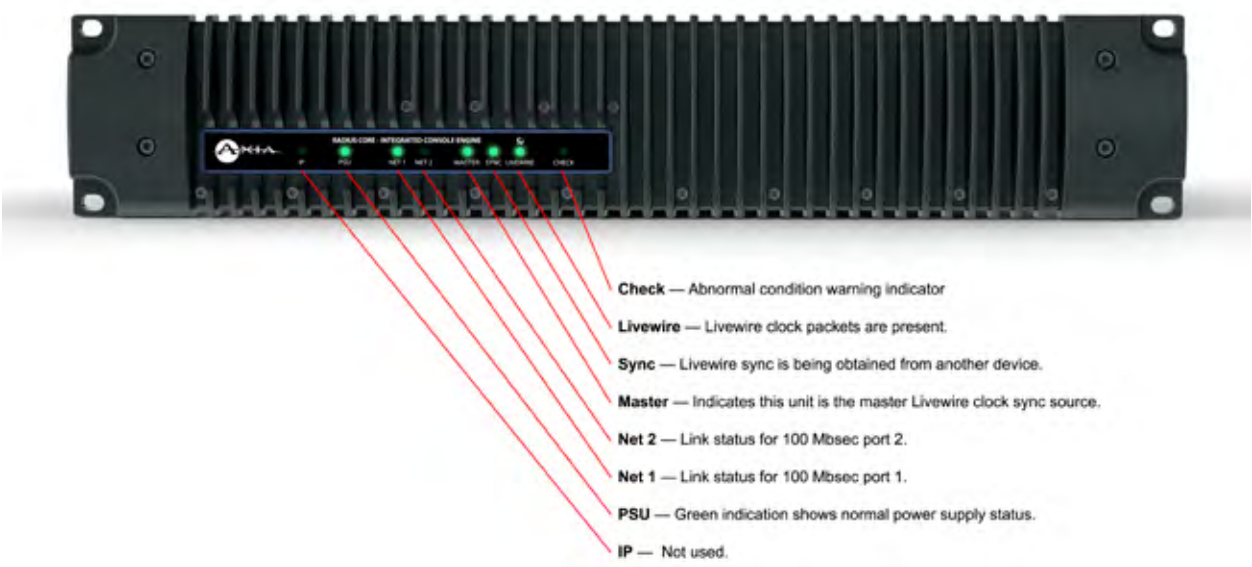
Check — Abnormal condition warning indicator
Livewire — Livewire clock packets are present.
Sync — Livewire sync is being obtained from another device.
Master — Indicates this unit is the master Livewire clock sync source.
Net 2 — Link status for 100 Mbsec port 2.
Net 1 — Link status for 100 Mbsec port 1.
PSU — Green indication shows normal power supply status.
IP — Not used.

Figure 1-1: Radius Core - Front Panel

CHECK

This is a general health indicator for the Radius Core. If this indicator is RED, immediately check the web pages for abnormal operating conditions such as excessive CPU temperature. Do not ignore this indicator. Note that it is normal for this indicator to flash during boot-up.

LIVEWIRE

This indicates that Livewire network traffic is present and multicast streams are properly advertised.

SYNC

If your Radius is part of a larger network, a solid GREEN indicates the Radius Core is synchronized as a slave to the Livewire network and that its internal clock is locked. Flashing GREEN indicates that sync packets are being received, but that Radius is not yet locked. It is normal for the **SYNC** indicator to flash for several seconds at startup. If it does not "lock" with a solid, non-flashing LED, this may indicate an incorrectly configured network switch. Check your Ethernet switch configuration or call Axia 24/7 Support for assistance.

MASTER

Solid GREEN indicates the Radius Core is synchronized as a Master clock to the Livewire network. This indicator will always be green for a standalone console. If you have any other Livewire devices, they should be reporting Sync or Slave. There should be only one Master clock device in a Livewire network at any given time. Figure 1-1 shows both Master and Sync illuminated for illustration purposes only.

> **Note:** The MASTER and SYNC indicators are not normally ON at the same time since the Radius Core will be either a Master or Slave. This principle applies to all Axia Livewire devices. Any Livewire network will have only a single Master.

NET 1 and NET 2

These indicators provide the status of the two 1000 Mbsec (Gigabit) NICs in the Radius Core. Note that the unit can be used with either or both ports connected to properly configured Ethernet switches or other Radius Core units.

When the Gig1 or Gig2 ports have active connections, the NET1 or NET2 lamps (respectively) will illuminate. If a network cable is connected to a Gig port and the corresponding NET lamp is unlit, a problem with the network connection is indicated. When a disconnected Ethernet link has been re-established, the NET LED will blink with a 50% duty cycle for about 5 seconds as the network activity is resumed.

Please consult Axia's Ethernet switch configuration guides for details when using Core switches in a larger network. Ethernet Switch guides may be found on our web site at AxiaAudio.com/manuals/.

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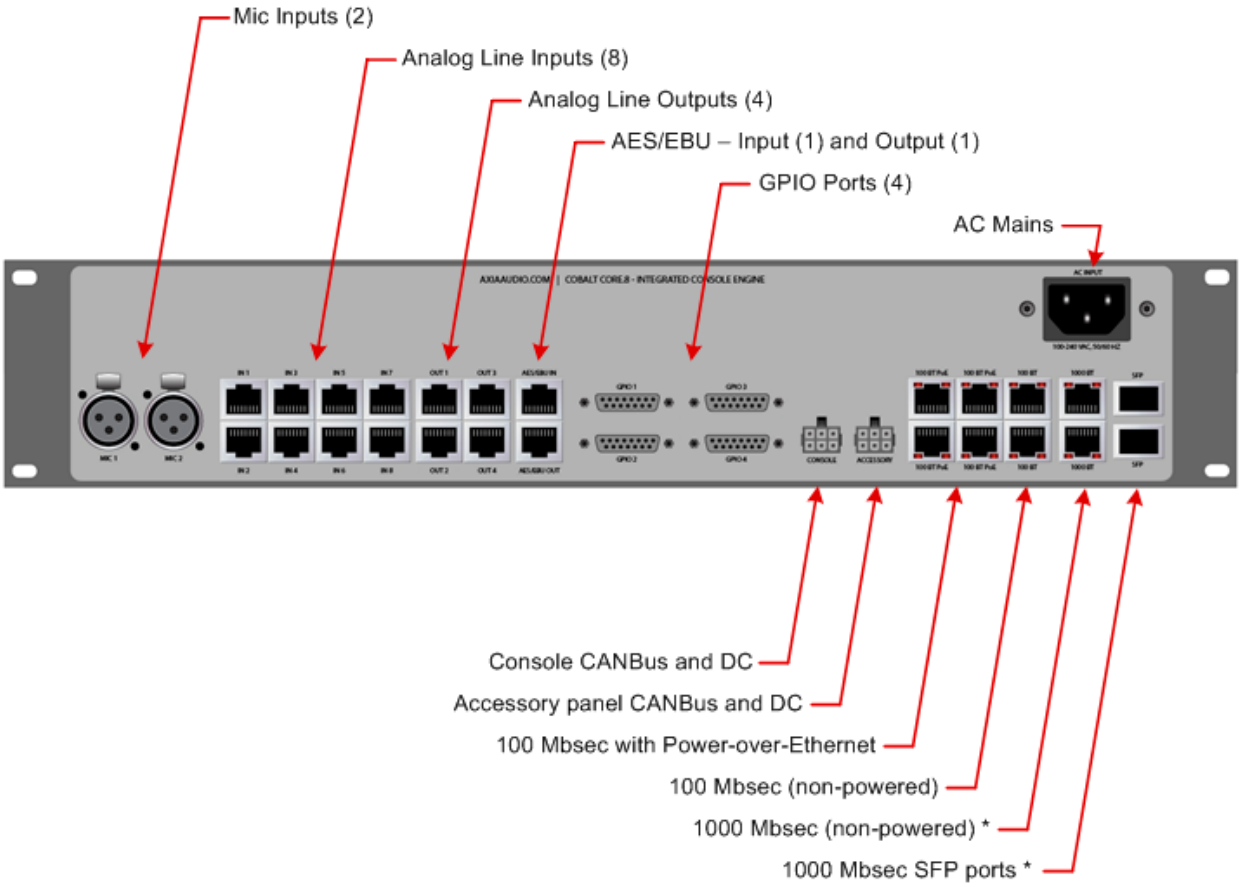
Mic Inputs (2)
Analog Line Inputs (8)
Analog Line Outputs (4)
AES/EBU – Input (1) and Output (1)
GPIO Ports (4)
AC Mains
Console CANBus and DC
Accessory panel CANBus and DC
100 Mbsec with Power-over-Ethernet
100 Mbsec (non-powered)
1000 Mbsec (non-powered) *
1000 Mbsec SFP ports *

Figure 1-2: Radius Core - Rear Panel Connections

PSU

The PSU indicator provides assurance that the Radius Core power supply rails are normal. Solid GREEN indicates normal power supply status.

IP

This indicator is currently not used.

Radius Core: Rear Panel

Power

- AC Mains: There is one standard IEC receptacle for your mains power for voltages in the range from 100 to 240 VAC at either 50 or 60 Hz.

Audio Connections

The Radius Core's rear panel contains all of the connectors used for audio I/O, Ethernet, GPIO, power supply and the connections to your Radius Fader Frames. Note that we use RJ-45 connectors for three purposes:

- Analog audio inputs and outputs
- AES audio inputs and outputs
- Ethernet (some with PoE - Power over Ethernet)

It is important to follow the proper conventions for wiring these connectors and ensure that the connector functions are not confused as damage may result if you plug a network cable into an audio input.

Analog and AES Audio

The connector pin functions shown in Figure 1-3 are the same for both the AES and Analog inputs and outputs.

The nominal audio level for analog line inputs and outputs is +4 dBm. This audio level will provide you with 20 dB of headroom.

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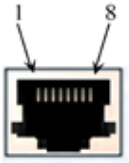

LINE and AES INPUT CONNECTORS	
Pin	Function: Analog/AES
1	Left Channel Input + /AES +
2	Left Channel Input - /AES -
3	Right Channel Input +
4	Not Connected
5	Not Connected
6	Right Channel Input -
7	Not Connected
8	Not Connected

Figure 1-3: RJ-45 Audio Connector Pinouts

Note: The well-labelled RJ-45 connectors on the Radius Core rear panel are used for different functions. Be sure to use them only for their intended purpose as described in this section.

Microphone Inputs

There are two XLR-F connectors on the rear panel for connection of microphone inputs. These connectors use the standard wiring convention for 3-pin XLR audio connectors. These connectors may have 48-volt Phantom power enabled via the Radius Core's web page configuration. Please take appropriate precautions if you connect anything other than a microphone to these inputs when Phantom power is enabled.

GPIO Connections

The Radius Core includes four GPIO ports, with pinouts in the same configuration as the GPIO ports on other Axia products. Please refer to **Chapter 5: Configuring GPIO** for the functions of these ports and wiring diagrams for their DB-15 connectors.

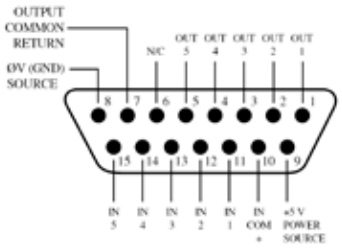

Figure 1-4: GPIO Connector Pin-Outs

In most cases, a GPIO port will be associated with a single source such as a CD player, studio microphone, profanity delay, *et cetera.*

Optional PathfinderPC software may also be used to monitor and control GPIO for advanced applications.

The interconnections to GPIO ports will vary depending on the type external equipment being used and whether or not voltage is supplied by an external device. **Chapter 5: Configuring GPIO** provides these details.

Ethernet Connections

The Radius Core has three types of Ethernet connections and it is important to use them as intended since the internal Ethernet switch is configured in a specific manner. The types of connectors available are:

- 100 Mbps PoE (Power-over-Ethernet) ports designed to be used with future Livewire accessories that will obtain their power directly from their Ethernet connections. These ports may also be safely used with standard, non-PoE devices.
- 100 Mbps Ethernet standard ports are used to connect PCs or other Livewire devices to your Livewire network. These devices might include additional Livewire nodes or PCs running the Axia IP-Audio driver.
- Two 1000 Mbps (Gigabit) ports are used for connections to other Radius Core units, or, in larget networks, for connections to an Ethernet Core switch. Note that these two ports are used as trunks and are not intended for connection to Livewire source devices such as Audio Nodes.

Radius CANBus Connections

Radius frames need only a single data / power connection to operate. The Radius Core includes two power/CANBus connections for use in conjunction with your Radius control surface and accessory modules.

Important: Like any electronic device, Radius can be affected by static electricity. Use of a personal grounding device is strongly recommended while handling the unit during installation.

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Your Radius Core **must** be connected to a grounded metal permanent wiring system or other equipment grounding conductor using the grounding screw located on the console chassis. For ground sources, we recommend, in order of preference:

- "Station Ground," the heavy copper strap found in the walls and floors of many radio studios.
- AC Safety Ground — the "3rd prong" of a nearby outlet.
- A bonded electrical conduit.

#12 AWG GREEN stranded wire is the *minimum* wire gauge acceptable for grounding the Radius.

Important: Grounding reduces the risk of electric shock by providing a "path of least resistance" for electric current. Improper grounding can result in a risk of electric shock.

Check with a qualified electrician if you are in doubt about how to properly ground this equipment. If your local electrical code prohibits the use of a Station Ground for this purpose, as described above, use the specific "Safety Ground" your local regulations mandate.

Console

There is one 6-pin Molex connector that is used to provide DC power (48 vdc) plus CANBus data to your Radius control surface.

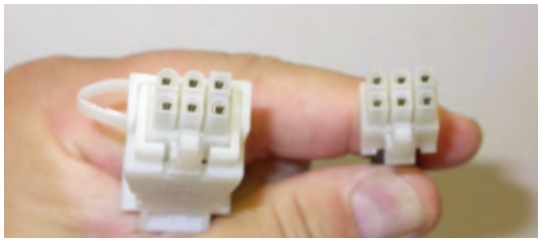

Figure 1-5: Detail of data/communication cable ends.

The Radius mixer must be connected to the CONSOLE port on the Radius Core using **only** the supplied 6-conductor cables. One end of the CANBus cable is fitted with a strain relief. Connect this end to the Molex connector on the back of the Radius Core. Connect the other, smaller connector to the Molex socket on the control surface.

Since the CANBus connection cables include power as well as data, there is a length restriction of approximately 50 feet. If you require longer cables than those supplied with your Radius, please contact your Axia representative.

Accessory

The other 6-pin Molex connector on the rear panel of the Radius Core is labelled ACCESSORY. This is for future planned accessories; it is currently not used.

What is CANBus? Controller Area Network (CAN) is a multicast shared serial bus standard, originally developed in the 1980s by Robert Bosch GmbH, for connecting electronic control units (ECUs). Although initially created for automotive purposes, nowadays it is used in many embedded control applications.

The CAN data link layer protocol is standardized in ISO 11898-1 (2003).

IP Address Configuration

To access System Settings from the Radius console, press-and-hold the **Meter** and **Clock** key for three seconds. The Studio display now shows the options and the Studio Monitor control is used to navigate, select options and change parameter values. Options available are:

- IP Address
- Subnet Mask
- Gateway
- Save and Reboot

To exit, select EXIT and press the Studio Monitor control.

Note: After making changes to any Radius Core IP address settings, you must reboot the unit for these changes to take effect.

"Gateway" settings on an Radius Core are not usually required; however, a valid gateway enables access to the Radius configuration remotely, from outside your network. A gateway also facilitates the use of an external NTP server. Enter the IP address of your master router (the one with external network connections) whenever you're given the option to enter a gateway address.

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Note: If your system employs an Ethernet core switch, the Ethernet core switch must have an IP address that is lower than any of your Radius units or Ethernet edge switches. This function of the IGMP standard ensures that the multicast querier is handled by the core switch if one exists. The querier will be passed off transparently to another switch should the core switch fail or be disconnected.

What's Next?

Take a break! You're done with initial set-up. When you're ready, continue to **Chapter 2: Radius Inputs and Outputs** to walk through the software configuration of the Inputs and Outputs on your Radius.

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Chapter Two:

Inputs and Outputs

In this chapter we'll discuss how to configure the inputs and outputs for your Radius console. First we will talk a bit about Profiles.

What's a Profile?

A Profile is a group of console parameters that are stored in memory for instant recall. The use of Profiles makes the operation of your Radius quick and intuitive. In the Radius console's design, there are two types of profiles: **Source Profiles** and **Show Profiles**. Show Profiles (discussed in detail in Chapter 3) contain all of the "snapshot" settings that would be associated with a scene or preset. This terminology is commonly used in the sound reinforcement or recording studio industries. Our **Show Profiles** keep track of monitor bus settings, fader channel assignments, jock preferences, feature lockouts and many other parameters.

You will use your web browser directed at the IP address of your Radius to set up Radius options including both types of profiles.

> ***Note:*** Axia web interfaces have been tested with Microsoft Internet Explorer, version 5 or later, but also work with other browsers.
>
> We also like Mozilla's FireFox browser, especially with regards to its "tabbed browsing" capabilities. With tabbed browsing, it's possible to bookmark the IP addresses of entire rooms worth of Audio Nodes, then recall them in tabbed groups with a single click.
>
> Java must be installed and enabled, and your pop-up blocker must be disabled. Java is available at http://www.java.com .

Source Profiles

Source Profiles are the first topic of this chapter. A **Source Profile** is used to identify the parameters that are unique to a local input or Livewire source as it applies to a specific Radius console.

One of the most repetitive and tedious parts of studio building has always been tying "data" to audio sources. Consider the wiring of line selector modules: you need a cable for the remote audio source, a set of wires for a "start" closure, another set for "stop" — even with TDM routers, an operation as seemingly simple as logic-follows-audio often turns out to be not so simple.

Mix-minus and IFB generation has also been a problem in the past. Too often it's a complicated process that requires air talent to correctly determine source and bus assignments at a moment's notice. How many times has unwanted audio made air thanks to this error-prone procedure?

Your Radius eliminates these headaches by automatically merging audio, logic and program data into a single, routable information stream. Since audio in a Livewire network is transported as packetized data, it's easy for us to "piggyback" other data along with audio. This is accomplished through the use of Source Profiles.

Radius Source Profiles also allow you to specify how network audio sources are to be used on each console in your facility. They assign modifiers to an audio input. Characteristics like Pan and trim. Even mix-minus source selections can be set in the Source Profile and automatically loaded whenever that source is assigned to a Radius fader.

The result of all this is that Radius is always presented with complete, consistent information about how you want a given audio source handled.

> ***Note:*** Source Profile settings do not transfer between studios. In other words, a mic in Studio A that has parameters applied through a Source Profile on the Studio A Radius will not have those settings automatically applied if that mic is loaded on the Radius in Studio B.

Source Configuration

We will now look at the input, or source section of the Radius.

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1. Open the web browser on your computer and point it to the IP address you previously set for your Radius.
2. You'll be prompted to enter a user name and password. The default user name is "user". Leave the password field blank.
3. Select **Sources** to proceed to the source setup page. You will now see the Sources section as shown in Figure 2-1. In this example, some of the sources have already been configured.

Note: The Radius has three different types of audio sources. Some audio sources originate with the rear panel input connectors and are used locally in the Radius. Other audio sources may also be obtained from other devices in your Livewire network. Lastly, audio sources (PGM1, PGM2, etc) are generated internally by the Core's DSP. First we will discuss configuring the local sources and making them available to Radius faders.

About Mics and Headphone Feeds: Radius provides you with several headphone options, all selectable from Radius monitor control section. The headphone output provides a feed intended for the Host (board-op's) mic position and includes the ability to monitor the Preview bus in headphones. Guests can talk to the host directly using the **Talkback** keys on their remote control panels.

The *Studio* headphone outputs send audio to studio talent and guest mic positions. Talent positions include a talkback channel that the board op can use to communicate using the **Talk To Backfeeds** function (guest positions do not have this Talkback channel).

An *individual headphone feed* can also be constructed to associate a dedicated headphone channel with a specific mic. This allows the board op to send individual, targeted talkback to a specific individual (using the **Talkback** keys) without other talent or guests hearing.

Creating a New Source

To create a new source, select a source type from the Source drop-down. Your choices are:

- **Operator** is the board operator's mic. It is the source mic for Radius TALK TO functions. Its logic mutes the CR monitors when ON.
- **Producer** is used for in-studio Producer's mic positions. It has associated GPIO logic which can operate TALK TO functions from a remote producer's panel. It also mutes CR monitors when ON.
- **CR Guest** is used for any other guest mic in the control room. Its associated logic port mutes the CR monitors when ON. It can have an individual headphone feed.
- **Studio Guest** is used for any mic located in a separate studio. Its logic mutes the Studio monitors when ON. It can have an individual headphone feed.
- **Line** is used for any line input audio source, analog or digital. A GPIO logic port can be used to provide machine start/stop pulses if desired.
- **Phone** defines this source as a hybrid or broadcast phone system input. A summed mono mix-minus is automatically provided.
- **Codec** marks this source as a codec. A dual mono mix-minus is provided; one PA feed (right) + one talent feed with talkback (left).
- **Computer Player** defines this source as originating from a PC. Associated GPIO logic can be used to send start/stop signals to a playout system.

Figure 2-1: Source Profiles screen

Once you have selected source type, click on the Create button to proceed to the Source Profile options.

Figure 2-2 shows the setup options screen for a typical Microphone source. Figure 2-4 shows a Codec source which has a few different settings and Figure 2-5 shows a Line source. The following sections describe these Source configurations in detail.

Source Type

This display-only parameter shows you the source type as selected in the previous step (Line, Codec, etc).

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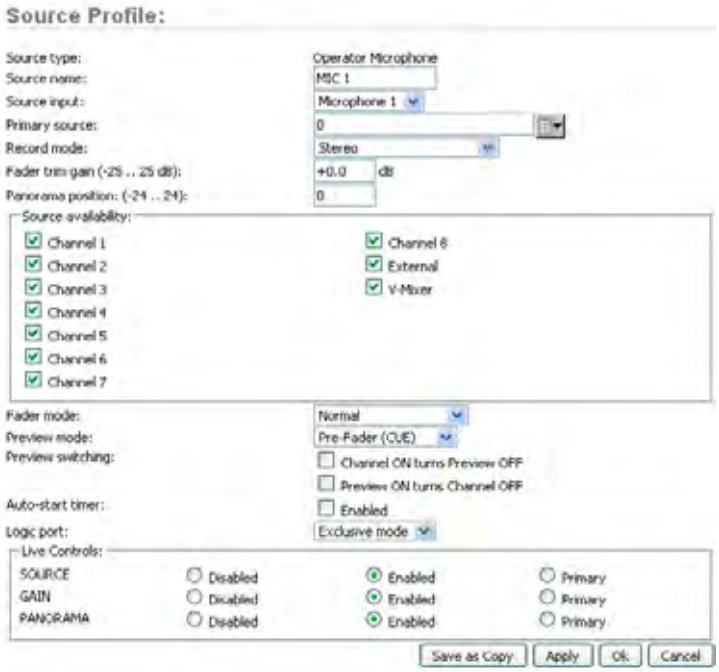

Figure 2-2: Operator Microphone Source screen

Source Name

Each Source needs a name so you will know what it is and where it is located. We suggest that you make this name as descriptive as possible. Don't make it too brief. As an example, don't use a name like **MIC-3** because that doesn't tell you enough. Use something more descriptive like **Studio-4 Mic-3** since this will tell you the location of the source as well as describe the type of source. Livewire names can contain any printable character and can be up to 24 characters long.

Source Input

This setting will associate this source with an audio input. It may be a local input in which case you would choose a local Mic, Analog or AES from the drop-down menu. This source may also be a Livewire source which means that it is a network source originating from some other Livewire device. When you choose Livewire as your Source Input, you will specify the source as described below.

Primary Source

This pop-up window is used to select the network Livewire source when Livewire has been selected above as your Source Input. Clicking the Browse button will open a new window like the one shown in Figure 2-3 where you will select your source from the list of those in the pop-up.

Signal Mode

Determines whether source will be treated as mono or stereo, and how.

- **Stereo** feeds incoming L/R signal to left and right channels of assigned bus(es).
- **Left** feeds incoming left channel to both channels of assigned bus(es).
- **Right** feeds incoming right channel to both channels of assigned bus(es).
- **Sum** creates L+R mono mix of incoming stereo source and feeds it to both channels of assigned bus(es).

Record Mode

The Record Bus of your Radius is a special variation of PGM 4 that is post-fader but pre ON/OFF. The mix of the Record bus is controlled by the **Record Mode** settings of each source. The options are:

- **Stereo** - delivers this source in stereo to the Record bus (default).
- **Sum to Left** - sums L+R of this source and delivers the sum to the Left channel of the Record bus.
- **Sum to Right** - sums L+R of this source and delivers the sum to the Right channel of the Record bus.

The last two options give you the ability to create a split Record bus which is common. The Operator's mic is usually recorded on one channel and all other sources are recorded on the other channel.

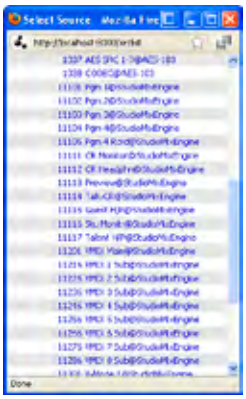

Figure 2-3: Livewire Source Select pop-up

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Figure 2-4: Codec Source screen

Fader Trim Gain

This setting Lets you specify an amount of from -25 to +25 dB that is applied to the signal just before the fader.

Panorama Position

Lets you preset pan settings for this input. Pan is variable in 49 steps, center being 0, far left -24 and far right +24. This setting can be adjusted on the fly by the operator if permitted by show profile settings.

Source Availability

These check boxes determine which faders (or other inputs) you'll allow this source to be assigned to. For example, if your source is a Control Room mic, and you only want it to appear on the left-most fader, you would check only the box marked Channel 1. You might want to make a CD player available on any fader, so you'd check the boxes for all faders.

Some sources – especially air monitors – you might wish to prevent from ever being assigned to a fader, so you would uncheck the boxes for all eight faders.

The two final check boxes, External and V-Mixer determine whether the board op will be able to assign this source to the CR and Studio monitor and headphone buses or the Virtual Mixer

To prevent a source from appearing on these inputs' selection lists, uncheck the desired box. Generally speaking, you'd want to restrict sources such as computers and CD players to fader assignments only while sources such as off-air receivers would be marked to appear only on the External monitor selection list.

Fader Mode

Defines fader start actions and states:

- **Normal** conforms to the US method of requiring talent to manually turn the channel on and off.
- **Fader Start** follows the common European standard of activating the channel and associated machine logic when the fader is raised from ∞.
- **Fader Start with Arm** works just like the Normal setting when the fader is not set to minimum position. When the fader is set to the minimum position, it can be in an Armed state.

Armed state is entered by pressing the ON Button when the fader is down. The armed state means that the channel will go to the ON state when the fader is moved up. When in Arm state, the channel OFF lamp will be illuminated. The channel ON lamp will flashing in "wink" mode - a long on-time and short off-time. For all fader modes, pressing the OFF button at any time will place the channel in the OFF state

Preview Mode

This permits a choice of whether **Preview** acts in CUE (pre-fader) or SOLO (post-fader) modes.

Preview Switching

There are two check boxes in this area. None, either or both may be checked. The options are:

- Channel ON turns Preview OFF

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- Preview ON turns Channel OFF

Auto-Start Timer

When this box is checked, the count-up timer will start when the fader is raised above its minimum position.

Logic Port

Allows you disable GPIO machine logic to this audio source device or enable it in two modes. **Exclusive** mode permits GPIO control associated with only a single fader. **Shared** mode allows more than one user to send ON/OFF/START signals to a source via GPIO.

A check box can optionally enable **GPIO Ready** mode. This allows devices such as CD players or tape machines to signal their READY state to the operator through the OFF button LED. If a device is cueing, its OFF lamp will not be lit. Tying the device's OFF lamp to the Ready command means that a machine must be cued and ready before the OFF lamp illuminates. If Enabled, the OFF lamp illuminates only when the fader is OFF and the Ready command is active on the GPIO. If Disabled, the OFF lamp illuminates normally whenever the fader is turned OFF.

Figure 2-5: Line Input Source screen

Feed to Source

If there is a backfeed associated with this source, this option determines what content is fed back from the console.

- **Disabled** - does just what it says and disables the backfeed.
- **Auto (Pgm-1/Phone)** feeds the Program 1 bus (minus the source) when this source's fader is ON, and feeds the Phone bus (minus the source) when the fader is OFF. Remember, the Phone bus is actually a special variant of Program 4. The Phone bus is prefader and pre-on/off to allow speaker-phone style operation thru the Operator's mic.
- **Program 1** feeds the Program 1 bus, minus the source.
- **Program 2** feeds the Program 2 bus, minus the source.
- **Program 3** feeds the Program 3 bus, minus the source.
- **Program 4** feeds the Program 4 bus, minus the source.
- **Phone** feeds the Phone bus, minus the source. Phone is a mono-sum, pre-fade and pre-on/off variation of Program 4

About the PHONE, RECORD and PROGRAM 4 buses: Radius PGM-4 mix bus is a special multi-purpose bus that does several things at once to make life easier for the operator.

Sources assigned to PGM-4 are also sent to the RECORD bus, a special post-fader, pre-on/off output that can be used to feed recording devices.

At the same time, sources assigned to PGM-4 are mixed and sent to the PHONE bus. PHONE is an internal bus which does not feed a separate output; instead, it picks up PGM-4 audio pre-fader and pre-On/Off and backfeeds it callers (minus themselves, of course) when the channel is **OFF** and **AUTO** is selected as the **Feed To Source Mode**.

Feed to Source Dim Gain

Normally, Talkback is sent over dimmed backfeed

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audio. If you want a hard interruption (Talkback interrupts Program audio), uncheck the **Enable** check box, which mutes the audio. But maybe your jocks would like to just have their program audio dimmed, and the Talkback come over it so they can still monitor air while they're being talked to. In this case, input a value by which the program audio will be dimmed. Adjustment range is 0dB to -30dB.

Live Controls

This group of radio buttons configures defaults and permitted functions for your **Options** control. One setting may be defined as the **Primary** which means that the Options control will vary as soon as you adjust the control. For example, if PANORAMA has been defined as the Primary function, simply rotating the **Options** control will Pan the source.

The other functions may be enabled or disabled as desired. Those that are **enabled** may be selected by depressing the **Options** control. When the control is pressed, the OLED display will list the options that have been **enabled** in this section of the configuration. Pressing the Options control selects the highlighted option and turning the Options control will adjust the parameter for that option. Depending on the enabled options, the operator will be able to adjust Source, Gain, Mode and Pan.

Local Outputs:
Output
Analog 1 CR Monitor
Analog 2 CR Headphones
Analog 3 Preview
Analog 4 Record
Analog 5 Disabled
Analog 6 Disabled
Analog 7 Disabled
Analog 8 Disabled
AES/EBU 1 Disabled
AES/EBU 2 Disabled
Apply
LiveWire Outputs:
Output Name Override Channel Mode
1 Program 1 Radius PGM1 31001 Standard Stereo
2 Record Radius REC 31002 Standard Stereo
3 Studio Guest HP Radius Studio HP 31003 Standard Stereo
4 Studio Monitor Radius St Montr 31004 Standard Stereo
5 V-Mix Radius vmix 31005 Standard Stereo
6 Auto Backfeed 0 Disabled
7 Auto Backfeed 0 Disabled
8 Auto Backfeed 0 Disabled
Apply

Figure 2-5: Line Input Source screen

Finally - remember to **save your work**. You may also "Save as Copy" if you intend to setup another similar source - you can save a few clicks this way.

Congratulations — now you know how to create Source Profiles! You can now bring up your new sources on the Radius faders.

Radius Outputs

Output Types

Now that we have configured the sources, let's look at the outputs of the Radius. Since the Radius is a network device as well as a console, you might expect that it has two kinds of outputs:

- **Local Outputs** - are the audio connectors on the rear panel of the Radius Core. These are used to connect Control Room monitors, Preview speakers, headphone amplifiers, record devices and, of course, your air chain equipment and processing.
- **Livewire Outputs** - are audio streams that are delivered to your Livewire network. These Radius outputs actually become network sources that can be utilized by other Livewire devices - more on this later.

Local Outputs

Your Radius Core has four stereo analog line outputs and one AES output. These outputs may be programmed to deliver any local sources or backfeeds and any of the Program or monitor feeds that are created by the Radius Core.

CAUTION - Once you have made a selection and applied it, audio is immediately delivered to the Core output. Be sure to turn down all level controls to avoid damaging your external equipment - especially your ears.

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Livewire Outputs

Since your Radius is a Livewire network device, it is capable of exchanging audio streams with your other Livewire devices. We have already seen how other external Livewire streams may be assigned to Radius faders. This section of the Radius configuration is where you choose the eight Radius outputs to be made accessible to your network as Livewire streams. This concept may be a bit difficult to grasp at first, but thinking of the Radius as a "normal analog" console may help. A normal analog output would be connected to the input of some other analog device. In our case, a Radius "output" is an audio source that may be connected to the "input" of another Livewire device.

Output - Using the drop-down menu, select the desired source that you want to deliver to your Livewire network.

Name Override - This the name that will be used when your output is "advertised" to the rest of your Livewire network. Simply provide a descriptive name that your can use to recognize this Livewire stream when it is viewed at other network locations. We suggest you include the studio name or location as a part of the name.

Channel - This section is where you specify the Livewire channel number. As discussed in the previous section, the channel number can be any number from 1 to 32,766. You may wish to follow the convention of using the last octet as a "base" for your Livewire channel numbers. For example, if the Radius IP address is 192.168.0.25, you may use 2501 through 2508 for the Livewire channel assignments of the Program and Monitor channels. The numbering convention easily includes the VMIX and VMODE sections and backfeeds.

Mode - The Mode function here is used to enable or disable the Livewire Output. Options are **Standard Stereo** or **Disabled**

V-Mixer and V-Mode

The Radius V-Mixer is an "auxiliary" mixer that can be used to mix five stereo sources and from this mix, create another new Livewire source that may be used like any other audio source.

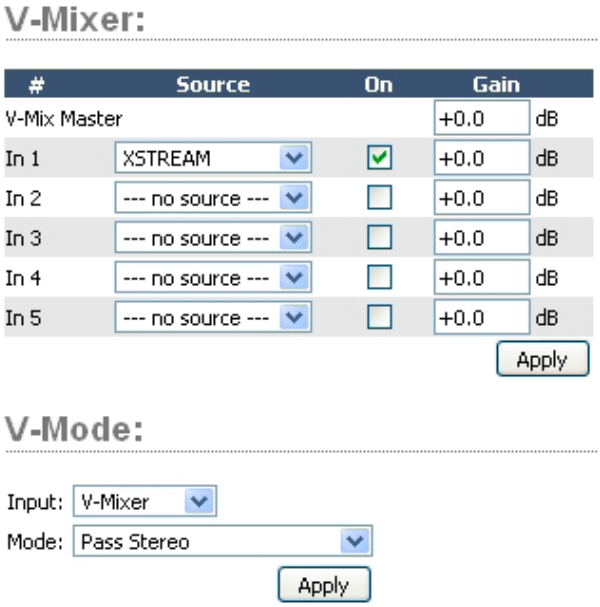
V-Mixer:
Source On Gain
V-Mix Master +0.0 dB
In 1 XSTREAM +0.0 dB
In 2 --- no source --- +0.0 dB
In 3 --- no source --- +0.0 dB
In 4 --- no source --- +0.0 dB
In 5 --- no source --- +0.0 dB
Apply
V-Mode:
Input: V-Mixer
Mode: Pass Stereo
Apply

Figure 2-7: Radius V-Mixer and V-Mode screen

V-Mode is a signal converter that allows you to manipulate some mono-mode sources.

V-Mixer and V-Mode outputs are assigned in the Radius Outputs section and delivered to your Livewire network. Once V-Mixer and V-Mode signals are a part of the network, they can be loaded on any local or networked console fader, or routed to any other network destination.

V-Mixer Controls

The V-Mixer works independently of the Radius console. In addition to static control of V-Mixer through the Radius web pages, Axia's PathfinderPC software can also be used to dynamically control V-Mixer and create mixing functions based on a variety of system-wide parameters. Please consult The PathfinderPC manual for details on that topic.

Even though the "V" in V-Mixer stands for "virtual", you can in fact perform a certain amount of "hands-on" mixing. Although you don't have physical control of the virtual faders or virtual **ON/OFF** switches, adjustments made to the V-Mixer take effect as soon as you apply them. While you may not wish want to operate an entire show this way, it's valuable to know that V-Mixer

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changes saved "on the fly" will take effect immediately.

Figure 2-7 shows the V-Mixer and V-Mode setup found in the Radius web pages. At the top of that page, you can see the V-Mixer submixer with its five stereo inputs. Below that you will see the V-Mode section. Let's look at the controls and see what they do.

The controls for the V-mixer's five input channels include:

- **Source:** From the drop-down menu, you may select any source that was previously defined with a source profile.
- **On:** Simply determines the ON/OFF status of the individual V-Mixer channel.
- **Gain:** This setting allows you to specify a gain setting for the channel that will control its mix level. The range is from -80 dB (effectively off) to +10dB. Note that you may also specify a **Gain** setting for the V-Mix master which is the output of the V-Mixer.

At the bottom of the submix section is an **Apply** button. Any changes you make will be saved when this button is pressed. Be sure to save your changes.

What does V-Mode do?

All audio within an Axia network travels as stereo Livewire streams. Most of the time, this is just what you need but in some cases, an application requires only one channel of an audio stream, or a mono sum. V-Mode lets you do this easily. Using it, you can strip, sum, or create custom audio streams from other available streams in the network.

V-Mode Controls

V-Mode allows you to perform special manipulations on a stream. The controls are as follows:

- **Input Selector** contains a drop down menu that allows you to select from a range of sources for each V-Mode input. You may choose from local Core sources including **PGM-1/2/3/4,** and **VMIX.**

- **Audio Mode** is the magic box that transforms your stereo stream into something else. You may choose from any of the following options:
  - » **Pass Stereo:** Passes the Left and Right channels as is, without modification.
  - » **Mono sum to L:** Creates mono sum and directs it to the Left channel only.
  - » **Mono sum to R:** Creates mono sum and directs it to the Right channel only.
  - » **Combine from left and right:** Creates a dual mono stream by summing the Left and Right channels.

Adding Backfeeds and GPIO

Since making a mix-minus and mapping contact closures is often done at the same time a new Source Profile is constructed, let's recap those procedures. Let's say you've connected a hybrid to Analog Input 4 of your Radius, named it "Studio 1 Phone". You've constructed a Source Profile for it, and you now want to give it a mix-minus backfeed and set up a GPIO contact closure for the "take" and "drop" functions. Here's how:

1. Make sure that, in your hybrid's Source Profile, the **Source Type** option is set to "Phone" and the **Feed to Source** is set to "**Auto (Program 1 / Phone)**".

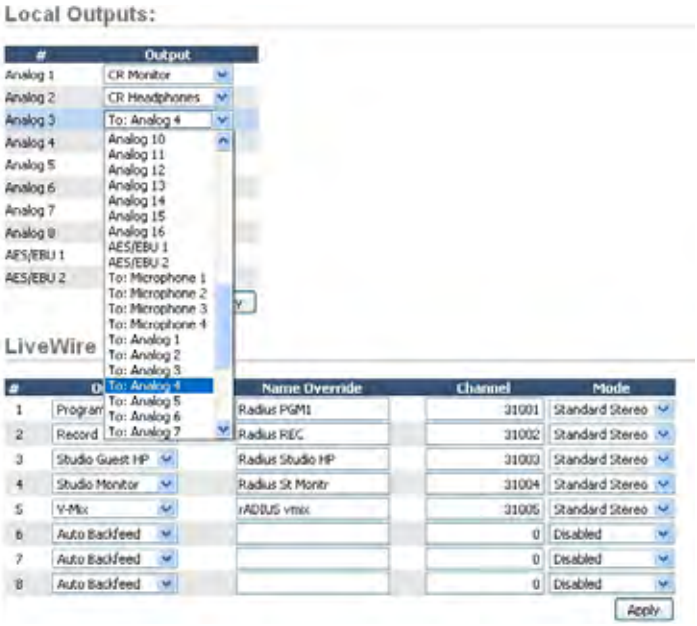

Figure 2-8: Picking a source from the list box

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2. Use your Web browser to connect to the Radius Outputs screen (Figure 2-8); we'll set up our mix-minus for output #3.

3. Just to the right of each **Local Output** name box , you'll see a drop-down selector. Click on the drop-down for output #3 and choose "to Analog 4" as shown in Figure 2-8. By simply choosing "to Analog 4", Radius knows that it is going to deliver a backfeed that is a mix-minus is Analog Input 4. The output chosen to deliver this backfeed is arbitrary - we happened to choose Output 3 for this example.

Automagic: If the hybrid source is loaded to an Engine and the engine is generating a mix minus source already, the list box will show TWO identical channels for the source — but one will have a description of "TO:" in front of it. Selecting this source automatically selects TO SOURCE in the TYPE box.

5. Click **Apply**.

6. Now point your browser to the Radius **GPIO Configuration**. You may also use a GPIO Node instead of Radius' built-in GPIO ports, just point your browser to that node's IP address instead.

7. The GPIO setup screen uses the same method of assigning devices to ports as an Analog Node. Pick a port to map your Phone channel's contact closures to, using the list tool, and click **Apply**.

The process is complete: you've made a Source Profile for your phone hybrid, created a mix-minus, mapped it to an audio output (destination), and associated a GPIO port for contact closures associated with the source.

To confirm, choose the **Options** key for any Radius fader channel and load your new Phone source. Now press the **Talk** key just below the fader; you'll see the Status change appear in the window above the channel to indicate that the operator's mic is talking to this telephone hybrid through it's backfeed channel.

Also, take another look at the GPIO screen on your browser. Note the green indicators that have appeared on the port representing your hybrid; switch the Radius channel **on** and **off** and you'll observe the active tally change pins.

What's Next

When you're ready, join us in **Chapter 3: Show Profiles** where we will show you how to setup your console and save the settings used for your shows.

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Mimicking nature,

streams combine to form anew

many become one.

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Chapter Three:

Operating Your Radius

Overview

Radius was designed to perform equally well in air or production studios. The clean interface enhances speed and accuracy without clutter or confusing controls.

In this chapter we'll first give you a high-level overview of Radius capabilities and then, detailed explanations of the various controls and functions.

Displays

Radius has built-in LED bar-graph meter displays. The meters can be configured to VU or PPM metering standards.

Radius has dedicated meter displays for Program 1 and Program 2 buses, and a multi-function meter. This meter can be toggled between Program 3, Program 4 and Monitor. The LED tally tells you what the meter is metering.

In addition to the meter displays, Radius has a time of day clock and a count-up timer.

Show Profiles

Radius can be completely re-configured, instantly, to suit different types of shows. By recalling any of four previously stored Show Profiles, talent can change board settings from a phone and microphone intensive morning show to a personality-based music show – or any other kind – at the touch of a button.

Show Profiles are easily toggled by pressing the **Profile** key located at the top of Monitor section. Construction and administration of Show Profiles is covered in **Chapter 4: Show Profiles.**

Sources, Channels and Faders

During the course of this chapter, we'll refer often to **sources** and **channels.** *These are not the same!* "Sources" are microphones, CD players, outputs from the playout system, telephone hybrids, etc.

"Channel" would normally be used when referring to Radius's linear faders and associated on/off controls, OLED displays, bus assignment keys, etc.

Sources are assigned to faders for use on-air or in production. During initial configuration, the Source Profiles are programmed by the engineer for appropriate logic and options (as outlined in previous chapters); afterwards, whenever that source is assigned to a fader, source logic follows.

Channel settings are instantly reconfigured whenever a source is selected, to accommodate the unique requirements of the source. For example, a fader controlling a microphone source also controls appropriate monitor mutes. A fader controlling a line source may send "start" and "stop" commands when the channel is turned on and off.

Sources such as an off-air feed can also be assigned directly to the monitors for auditioning without being assigned to a fader.

Mix-Minus

Mix-minus setup, especially for live broadcasts, has always been one of the most confusing aspects of running a radio console. Radius makes mix-minus easy by automating it. The operator never has to worry about sending a source back to itself and getting feedback — it just can't happen!

Several fixed and auto-switching mix-minus choices are possible, and are configured for each source when Source Profiles are defined by the engineer (see **Chapter 2** for details). Once this setup is done, no further tweaking is needed; the operator simply uses the source. A fader's status display tells operators when a particular source has a mix minus output, and even tells them which audio mix is being backfed.

Radius will generate as many mix-minus outputs as

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there are faders, each with its own automated mix minus feed. 8 faders = 8 possible mix-minuses!

GPIO

Chapter 2 explained how sources can have associated GPIO (General Purpose Input/Output) control associated with them. Radius has four built-in GPIO ports, but external GPIO interfaces can be connected to the Ethernet switch built into Radius Core, or via a Livewire network connection (more details in **Chapter 7**), so you can locate the actual machine connections either physically close to source equipment or in a central equipment room.

During installation, the engineer configures sources to accept and provide logic commands by selecting various options. For example, studio mics may be set up to automatically mute the studio monitor speakers and illuminate an on-air warning light when turned on. The same sources can be configured to accept GPIO logic inputs from buttons, so that each studio guest can have their own **On, Off, Talk,** and **Mute** keys.

Line sources can be configured to receive *start* and *stop* commands from the Radius as well. Start/stop messages can also be passed, via Ethernet, to a computer-based playout system.

Software

The Radius operating system uses a high-performance embedded Linux kernel to help ensure bullet-proof, 24/7 operation. Software updates can be obtained from Axia via the Internet and applied by the station engineer. Axia Support recommends performing backups or making printouts of Show Profiles and other configuration data before applying new software updates.

Callouts and Operation

This section provides a "bird's-eye view" of the Radius, with full descriptions of its controls and their functions.

Although Radius consoles have a fixed number of faders, you are not limited to that number of sources. You may also assign any networked source to any fader.

The Radius control surface is laid out in a conventional manner with the monitor and ancillary controls located on the right side of the frame. Let's look at each of these sections in detail.

1. **Channel OFF Switch:** You know this!

2. **Channel ON Switch:** When the channel is in fader-normal mode the **ON** key turns the channel on and the **OFF** key turns it off. The channel **On** and **Off** functions also provide logic (start and stop pulses, monitor muting, etc.) specific to the source type.

   For example, if a control-room mic is assigned to a fader, the CR monitors and Preview speakers will mute when that fader is turned on and when CR microphone sources are Previewed. If a fader is hosting a microphone source, pressing the **On** key when the fader is already on mutes the channel, providing a "cough" function, until the **On** key is released.

3. **SOFT Key:** Each channel has a Soft Key. The OLED display tells you what function the soft key is currently controlling. The default action depends upon the type of input assigned to that fader, which is defined in the Source Profile. Current functions of the key are Mute, Talkback and phone hold. This key is also used to exit Channel Options edit mode.

4. **PREVIEW Key:** allows the operator to listen to sources before they air. Radius' **Preview** bus allows auditioning in full stereo. Individual source profiles can change Preview to Solo (post-fader) operation.

   The **Preview** key acts like a latching switch.

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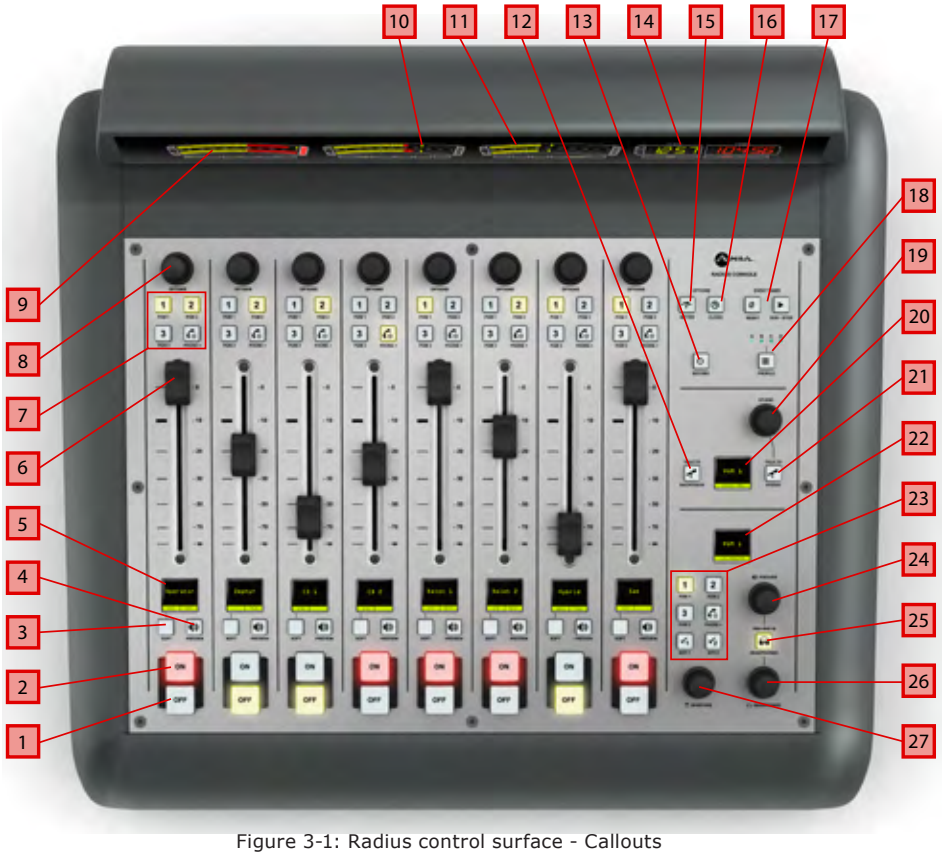

Figure 3-1: Radius control surface - Callouts

Momentarily pressing any channel's **Preview** key assigns that channel to the Preview bus; pressing it again removes the channel from Preview. To enable an operator to quickly listen to various sources, the Preview function can be interlocked: pressing **Preview** on any channel will remove any other channel from Preview mode.

If you want to preview multiple channels, pressing and *holding* any **Preview** key temporarily disables the interlock; other sources can be added to the Preview mix by pressing them. Conversely, channels can be removed from a multiple Preview selection by pressing and *holding* any lit **Preview** key and deselecting the individual channels you wish to remove from Preview.

A *momentary* press of any lit **Preview** key will remove all channels from Preview.

The interlock option for Preview can be disabled in any Show Profile's Monitor section so that users can gang preview sources without the need to press and hold **Preview**.

The audio from the Preview bus feeds any speakers dedicated to Preview, and can also be routed to the operator's headphones and CR monitors if desired. Preview speakers will be muted whenever and CR microphone is **ON**, or is assigned to the Preview bus.

5. **Channel Information Display:** This OLED display interacts with the **Channel Options** control and the **Soft** key. It is context-sensitive and can display many different parameters relevant to each fader. Normally the display will show the name of the source that is applied to the fader ("Zephyr"), and the function assigned to the **SOFT** key ("TALK").

When fader channel parameters are being edited, this display shows gain and pan settings, mode,

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backfeed selection and other parameters that will be dependent on the type of source.

When viewing these menus, the **Options** control is pressed to select the function, and rotated to set the value.

Just below the Source name is a status line. It shows the status of a channel. It is displayed in medium font. In the previous example, it is "[TALKBACK]."

Here is a list of the different status displays and their meanings:

» [TALK TO CR] - means that this channel is talking to Control Room. Used on Guest Microphone and Codec source types

» [OP TALKING] - means that this fader is an operator mic and it is currently in use for **Talkback** rather than feeding to Program buses

» [MUTED] - means that this channel is muted

» [TALKBACK] - means that this channel is being talked to, i.e., the talk bus is fed to the backfeed stream for this fader.

» Src-label - If the source has an associated label, it will be shown. Source label is normally used by playout systems to indicate song name and/or important information. Available only for Livewire sources.

» Listen Only - The source was loaded in listen-only mode. No GPIO control is available for this source's originating device.

» Used EW - The source is currently in use elsewhere and is in listen only mode. As soon as source is freed, it will be automatically reloaded in Full Control mode.

» Src Load Err - An error occurred during source loading. Re-load the source.

» No Backfeed - An error occurred during backfeed allocation for the source. For example, no resources are available to allocate backfeed stream.

» Src Waiting - means that a new source is waiting to be loaded. This state is accompanied by a blinking **OFF** key. Turn the channel **OFF** to load the new source.

6. **Fader:** The fader controls the volume of the input source. There are two modes for the fader: **fader-start** and **fader-normal:**

- When the channel is used in **fader-start** mode, pushing the fader all the way to the bottom of its travel turns the module off, and sends appropriate logic commands to source equipment. Moving the fader up turns the channel on and sends logic commands. In **fader-start** mode, the **On** and **Off** keys act more as indicators of channel status; the **On** key will not turn the channel on if the fader is at the bottom of its travel. Pressing the **Off** key while moving the fader up suppresses turning the channel **On.**
- In **fader-normal** mode, the on/off status and start/stop commands follow the channel ON/OFF switches independently of the fader position.

7. **Program Bus Assignment Keys:** Each channel is assigned to any of the three main output buses by selecting any (or all) of these program bus keys. Generally, PGM-1 is the main on-air bus and other buses are used for production or other programming requirements. All program outputs are post-fader and post on-off function.

The fourth bus functions as a regular output bus, but any sources feeding PGM-4 are also sent to a special **Record** bus that can feed a dedicated recording device, as well as to the **Phone** bus that sends mix-minus audio to any phone callers. These buses operate independently but simultaneously, allowing jocks plenty of operational flexibility.

» **PGM-4** bus audio output is post-fader, post-ON/OFF.

» **PGM-4 Record** bus audio output is post-fader, pre-ON/OFF.

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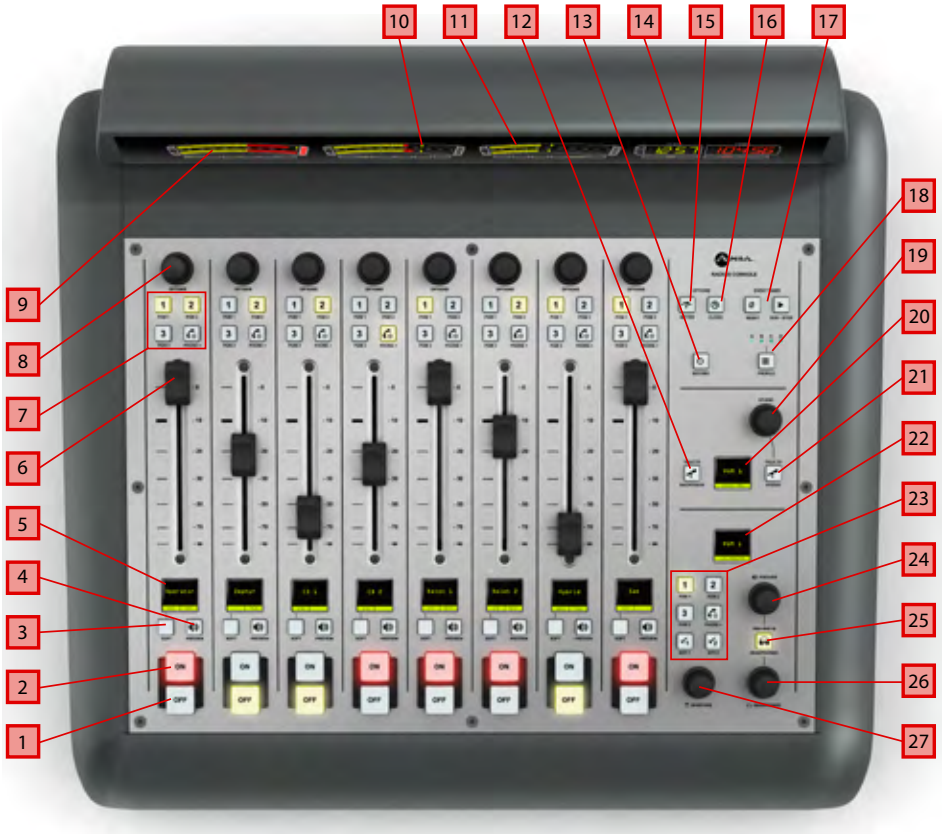

» **Phone** bus audio is sent to callers pre-fader, pre-ON/OFF.

8. **Channel Options Control:** This control operates in conjunction with the OLED fader display. Pushing the rotary encoder selects OPTIONS mode, displaying various choices on the OLED beneath. Rotate to select the desired option, and then push to "take." Once an option has been selected, you can rotate the control again to adjust the value of that parameter (if any). Press the **Soft** key to exit.

In the Radius Core's Web setup, you can also assign a default function to the rotary encoder. Doing so allows adjustments to fader options, gain trim, input gain trim, pan, or input selection to be made just by rotating the knob.

9. **PGM 1 Meter:** Always displays the output level of the Radius Program 1 bus.

10. **PGM 2 Meter:** Always displays the output level for the Program 2 bus.

11. **Soft Meter:** displays the output level of a bus that may be selected by the operator by using the Meter Select key (14). Options for this meter are: PGM 3, PGM 4 or Follow Monitor. The 3rd meter also has a "Check" warning tally to warn your engineer of an abnormal state that should be addressed.

12. **TALK to Backfeed:** this key allows the operator to instantly talkback to any phone or codec source(s) by pressing a single key. Pressing the TALK to Backfeed key routes the board operator's mic to all existing backfeeds. i.e.: you can talk to a codec or phone caller's mix-minus feed by simply pushing TALK to Backfeed.

13. **Record:** places the Radius in **Record Mode.** Monitoring (CR Headphones) will switch from the current source to the Record Bus, The 3rd meter will display the Record Bus levels. Control Room Monitors and Preview speakers will be muted. Pressing a **Record** second time releases **Record**

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Mode and restores Radius to its previous state.

14. **Clock and Timer Display:** This display includes both the Clock and the Timer. The Clock section includes an NTP indicator that tells you when the Radius synchronized to an external NTP clock reference. The Timer section displays the count-up timer that may be controlled by fader-start or by the Timer controls on the Radius.

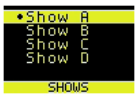
15. **Meter Key:** This key toggles the Soft (third) meter between PGM 3, PGM 4 and MONITOR. The default is PGM 3. If you look at the backlit labels on the bottom of the 3rd meter, you will see the MONITOR label that lights up when this meter option is set to "follow monitor".

16. **Clock Key:** Pressing this key toggles between AUTO and MANUAL modes for the count-up timer. AUTO allows it to be reset and started by any source starting that has it's timer start option enabled. MANUAL ignores those starts and follows the RUN/STOP/RESET keys only. Note that even if you are set to AUTO, the run/stop/reset keys still work.

To set the clock, push and hold the **Clock** key for 6 seconds to enter clock adjust mode. The **Clock** key will illuminate indicating that clock setup is active. Release the **Clock** key and press RUN-STOP to set the clock display forward, and press RESET to set it back. Press the **Clock** key again to save the current time and exit clock adjust mode.

If NTP is active and enabled (via web page configuration), the time cannot be adjusted manually. **Record:** places the Radius in **Record Mode.** Monitoring (CR Headphones) will switch from the current source to the Record Bus, The 3rd meter will display the Record Bus levels. Control Room Monitors and Preview speakers will be muted. Pressing **Record** a second time releases **Record Mode** and restores Radius to its previous state.

17. **Timer Controls:** Pressing **RESET** zeroes the count-up timer. Pressing the **RUN/Stop** key manually starts or stops the count-up timer. The **RUN/Stop** key illuminates while the timer is running



18. **Profile Key:** This key allows you to select the active Radius Show Profile from a group of four preconfigured profiles. The Studio OLED display tells you which profile (A, B, C or D) is currently active. The Studio Monitor knob is used to navigate this screen. To exit this screen, press the **Profile** key. You can also push and hold the profile key for 5 seconds to reload the current profile.

19. **STUDIO Monitor Control:** "Studio" is defined here as another room separate from the Control Room, such as a talk studio or news booth. Pushing this control selects the Studio Monitor source; rotate it to adjust the volume of the Studio Monitor speakers. Note that the Studio monitors will be muted whenever any Studio Microphone is turned ON.

20. **STUDIO Monitor Display:** This OLED display indicates the volume and source that is being delivered your STUDIO monitors. This may be changed with the **STUDIO** control as described above.

21. **TALK to STUDIO Key:** This push-to-talk key allows the operator to communicate with all Studio guests simultaneously. The operator mic is routed directly to Studio Headphone and Studio Monitor outputs while this key is pressed.

22. **Control Room Monitor Display:** This is a context sensitive OLED display that indicates the volume and source that is being delivered to your Con-

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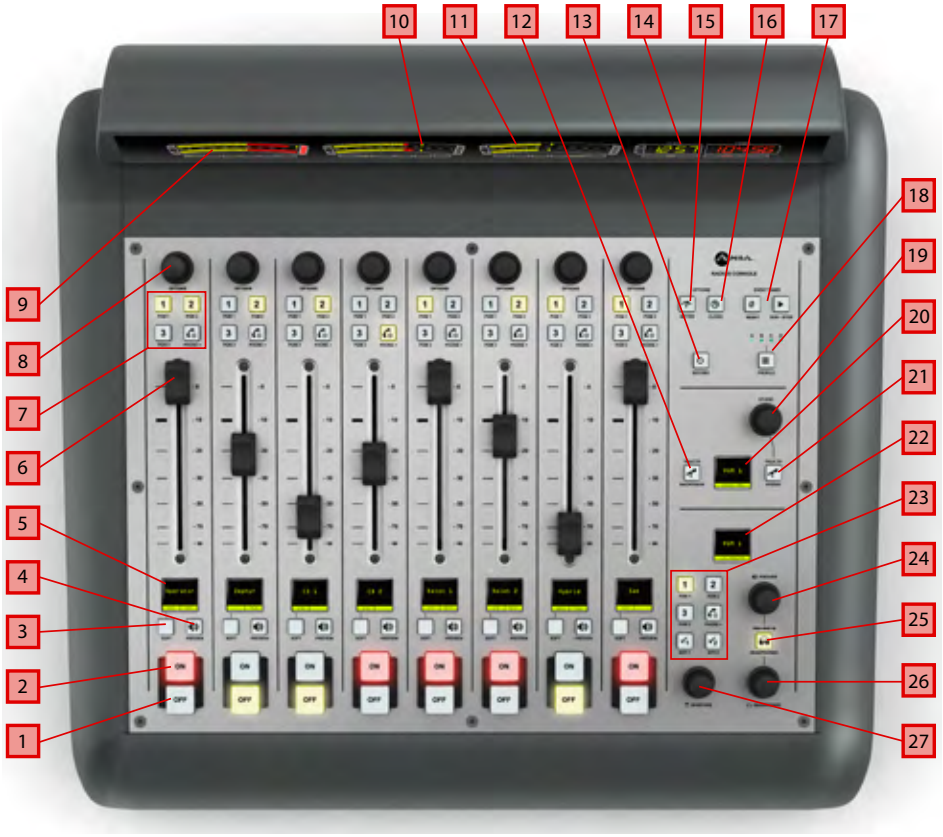

trol Room monitors, CR Headphones and Preview.

23. **Control Room Monitor Select Keys:** This group of keys selects the source that is delivered to your control room monitor speakers. The operator may select from PGM 1, PGM 2, PGM 3, PGM 4 or two pre-designated external sources. The external sources are typically used for off-air receivers and are specified in the web page configuration.

External 1 and External 2 have another function, in addition to their normal monitor-select action. If you press and hold one of these keys for 2 seconds, you will see that the Studio OLED display and Studio Monitor knob may be used to assign a source on-the-fly to that specific External Input. Press **Ext 1** or **Ext 2** again to exit this Select mode.

24. **PREVIEW Level Control:** Adjusts the volume level of the Preview output connected to your external powered **PREVIEW** speaker. Note that the **PREVIEW** speaker will be muted whenever any CR microphone is **ON** or when you are in **Record** mode.

25. **PREVIEW in HEADPHONE Key:** The **H/P PREV** key controls whether audio from the **Preview** bus will be fed to the board op's headphones, and if so, in what manner. Pressing this key toggles through the following options:

» **Off:** The operator will not hear Preview audio in the headphone channel.

» **ON (Stereo):** audio from the Preview channel overrides the **Headphone Monitor** selection and is sent to both the left and right channels of the operator's headphones.

26. **HEADPHONE Level Control:** Rotate to adjust the Headphone volume. Headphone source follows CR Monitor selection.

27. **MONITOR:** Adjusts the level of your Control Room Monitors. This output will automatically MUTE whenever any Control Room microphone is **ON** or when Record Mode is active

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System Settings

To access System Settings from the Radius console, press and hold the **Meter** and **Clock** keys for three seconds. The Studio display shows the options and the Studio Monitor control is used to navigate, select options and change parameter values.

Options available are:

- » **IP Address**
- » **Subnet Mask**
- » **Gateway**
- » **Save and Reboot**

To exit, select **EXIT** and press the Studio Monitor control.

Source-Specific Channel Controls

Some fader functions change their behavior to suit the type of source assigned to the fader. A Guest Mic input has unique functions different from those of a Codec input, and so forth. In this section we'll explain the channel functions unique to each type of source input.

Control Room Operator Mic Channel

The Control Room Operator microphone is the board operator's mic. It's always located in the Control Room, so activating a channel designated as the CR Operator mic affects CR Monitor speaker and Preview speaker muting.

When the board operator pushes a **Talkback** key, all output bus assignments for the Control Room Operator microphone are temporarily muted, while his mic's audio is routed to the requested **Talkback** destination. Releasing the key returns the channel outputs to normal.

Pressing the channel **ON** key for the CR Operator Mic while the channel is already on will mute the channel until the key is released, acting as a "**cough**" function for the board operator. On-air status and speaker mutes are unaffected by this action. The CR Operator microphone channel will mute the CR Monitors (Monitor 1) and the Preview speakers whenever the channel is **On,** or if **Preview** or **Record Mode Engage** keys are selected.

Producer Microphone Channel

The Producer's microphone is meant to be used at a position within the Control Room where a morning show or talk show producer would normally sit. This mic type mimics the operation of the Control Room mic described above, but does not include the "cough" function; also, a Producer mic can use Talkback to talk to the board op, but cannot talk directly to the Studio talkback channel.

Control Room Guest Microphone Channel

There are often microphones in the Control Room other than the board operator's; i.e., an in-studio guest or perhaps an announcer located physically near the board op. These microphones also control the muting of the Control Room monitor speakers.

Control Room Guest Microphone channels will mute the Control Room speakers and the Preview speakers whenever the channel is **ON,** or if **Preview** key is selected.

Studio Microphone Channel

Another type of microphone channel is a Studio Microphone. This mic is typically located in a location separate from the control room — perhaps a voice-over booth, a talk studio separated by glass, or a news booth.

Logic for Studio Microphone channels is identical to that of the Control Room Guest Microphone channel. The **Studio Monitor** speakers mute when a Studio Microphone channel is turned on, or if it is assigned to the **Program-4/Record** bus.

External Microphone Channel

This type of microphone operates identically to a studio or CR guest microphone except that is does not incorporate any monitor muting functions.

Line Channel

Line-level devices like CD players, tape players, etc., use the Line Channel profile. The Line Channel is ste-

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reo, but the user can change the input mode to mono, left only or right only by using the **Pan & Mode** key in the Master Module's **Channel Options** section.

Line channels have a variety of remote logic functions available at the GPIO interface. **ON** and **OFF** keys can be remotely controlled, as well as the **Preview** key. Each function has a corresponding lamp driver.

You'll also find a **Ready** command which will remotely illuminate the **OFF** lamp on the Radius channel and activate the GPIO **OFF** lamp driver, allowing source equipment to signal the operator with either a steady-state **OFF** key (indicating source ready) or a flashing **OFF** lamp (indicating source next). Refer to the *GPIO Node User's Manual* for information on interfacing to the GPIO's inputs and outputs.

Phone Channel

The Phone Channel profile is used, naturally, for telephone hybrid audio sources. Each Phone Channel has its own discrete Feed-to-Source mix-minus output.

Each Radius can accommodate as many unique, automatically-generated mix-minus feeds as it has faders. If the Feed-to-Source mode is set to "Auto", (the most common option - see **Chapter 2: Inputs and Outputs** for the section entitled "Source Profile Options"), phone callers hear the output of the **Program-1** bus, minus himself, when the channels they're assigned to are **ON**. When the caller's channel is **OFF**, their channel's **Talkback** key is lit and they hear the Board Operator's mic routed to their **Talkback** feed.

Phone feed-to-source outputs can also be locked to backfeed only **PGM-1, PGM-2, PGM-3** or **PGM-4**, using the options provided when constructing the Source Profile for your hybrid. This automatic switching mode makes it very easy to prep callers and then seamlessly take them to air with a minimum of key pushes.

Codec Channel

The Codec Channel is used for codec sources. Like the Phone Channel, each codec source is brought up on its own channel, and an individual Feed-to-Source (mix minus) output is provided for each codec. Codec Channels can be turned on and off remotely and provide on/off status to a remote logic device. **Remote Talk** and **Remote Mute** are also provided, allowing talent to take control using their own On/Off/Mute/Talk panels.

Talk enables remote talent to communicate with the board op using the **Talkback** bus. When **Talk** is activated, all outputs for that channel are muted and the source audio is fed to the **Talkback** bus; the channel's **OLED** displays TALK to CR. When the user stops talking, the channel returns to normal.

When the operator uses the Softkey to talk to the phone or codec source, the OLED displays TALKBACK.

When the remote mute function is activated, the OLED displays MUTED and all the channel outputs are muted until the command is released.

The mix-minus output (configured in the Source Profile section) normally feeds **Program-1** audio to the remote, but can be switched to **Program-2, 3, 4 or Phone**. The Feed-to-Source output on a Codec Channel is a dual mono output, occupying both sides of the stereo pair. The Left output is normally used for remote talent's headphone feed, and is interrupted by **Talkback** audio when the board op talks to the remote. The Right output sends uninterrupted program audio (without talkback) for use as a PA feed.

Remote Control

Console channels can be turned on and off remotely, and will provide on/off status to a remote logic device. **Remote Preview** is also provided, as are lamp drivers for **On, Off** and **Preview. Start** and **Stop** pulses can be sent when the channel is turned on or off, respectively. See **Chapter 5: Configuring GPIO** for detailed information interfacing to GPIO inputs and outputs.

What's Next

Now that you're familiar with Radius controls, let's learn to tap the power of Show Profiles.

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Do digital streams

dream of electric DJs?

Just Bradbury knows.

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Chapter Four:

Show Profiles

If you've read **Chapter 2: Inputs and Outputs**, you already have a good understanding of Source Profiles. In the same way that Source Profiles allow you to determine how an individual channel is configured when a source is loaded to a fader, Show Profiles let you build configuration files that determine how the entire console behaves, which sources are loaded, and what channels they appear on — "snapshots" that you can load at will to recall baseline show settings.

For instance, you could use Show Profiles as a "return to zero" command. Radius is highly customizable; jocks can might load new sources, change EQ or make other adjustments during their show. A Show Profile can reset the console to its default state for the next operator.

Or, you could use Show Profiles to define different types of broadcasts – one for the morning show, one for talk segments, one for musical guest interviews, one for unattended operation – that need only to be loaded to instantly optimize the board for use with that situation.

You can save and recall up to four unique Show Profiles to customize your Radius for different users or shows.

Creating A Show Profile

Build A Show

Radius' Web interface, which you've used to construct Source Profiles, is also used to help build Show Profiles, so make sure a computer connected to your Radius Core or Livewire network is within easy reach.

Your Radius is shipped from the factory with a default Show Profile. This loads the first time you turn on your Radius; it provides the "template" you'll use to construct custom Show Profiles. You've no doubt been exploring your Radius, so you might want to reload **Default Show Profile A with** the **Profile** key located in the Radius Master section. Simply press and hold the **A** key and, after a few seconds, the default profile will load, resetting the mixing surface to its factory state.

The first step in building a Show Profile is to set up Radius using the Channel Options, Monitor, Headphone, Studio and other board options as described in **Chapter 3**. Get started by assigning a source to each Radius fader using the **Options** knobs at the top of each channel.

After you've assigned a source for each fader, you can configure the rest of the Channel Options:

- Gain,
- Pan / Balance,
- Feed to Source.

When you've configured all Channel Options, continue on and make monitor and headphone selections since these are all saved in Show Profiles as well.

When you've got the board configured to your satisfaction, proceed to the next section.

Capture It!

It's now time to save your configuration as a Show Profile. Enter your Radius' IP address in your Web browser. When the Main Menu appears, choose **Shows** on the sidebar menu. This is where you'll come whenever you need to capture, construct or administer a Show Profile.

Under the Show Profiles list, you'll find four profiles listed with three options for each Show Profile:

- **Edit.** Choose this option to manually edit any existing show profiles.
- **Capture.** Takes a "snapshot" of the current Radius console state and saves it for recall. This is the option we'll use in just a moment.
- **Reset.** Choose this option when you want to construct an entire Show Profile completely from scratch and reset all parameters to their default state or value.

You've already set up your Radius, so now you can just "capture" your settings for further use. Choose **Cap-**

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Figure 4-1: Radius Show Profiles screen

ture from the menu. Radius allows you to capture up to four Show Profiles which are saved as A, B, C and D. You may also give meaningful names to these Show Profiles as described in the next section.

There may be a few more options you'd like to tweak, since not all options are accessible from Radius console itself. Let's go a little deeper into Show Profiles to find out exactly what's available to you.

Suggestion: As part of Radius training, show your talent how to use the Channel Options to assign sources and set monitor options. Let them set up the board exactly the way they want it, then use the Capture command to save their work. This way, talent gets the satisfaction of personalizing their own Radius configuration — and saves you the work!

Show Profile Options

Although the ability to create Show Profiles using the "capture" function is powerful, there are a few more options available exclusively via the Radius Web interface. You will probably find that a combination of console setup, capture and edit works best.

When you select any Show Profile, you will enter the edit screen as shown in Figure 4-2. Notice that there is a section for your Radius faders, as well as sections for Phone, Monitors, External Inputs and some general User Interface settings. At the very top of the page is the **Show Name**. Use this to give the Profile a descriptive appellation

We will now look at the individual Show Profile settings in detail.

Fader Channels

This section has settings for each fader channel. When this specific Show Profile is loaded, these settings will be applied.

- **SOURCE:** You can change the source using the drop-down box. The name of the source you assign will be displayed on the OLED below the fader.
- **ON/OFF:** Specifies the channel ON/OFF state when the profile is loaded.
- **PGM1/2/3/4:** Determines the default bus assignments for this channel when the Show Profile is loaded. When a check-box is selected, the source will be assigned to the corresponding bus.
- **Group Start:** There are three possible settings:
  - » **Independent** - this fader is not associated with any other fader and operates independently
  - » **Master** - is used to select one fader as the master for group starts.
  - » **Slave** - is used to select one or more faders as Slaves that will follow the action of the fader that has been designated as the Master.

Phone Connection

This section contains the settings for connection to your Radius6 or Telos VX phone system. (Of course, you can also connect your Radius to any stand-alone hybrid by using the inputs and outputs on the Radius Core, but the settings contained in this section function only with Radius6 or VX phone systems.)

Note that the Radius phone source profiles contain settings that are used to specify individual backfeeds and hybrid channels for *all* types of phone systems.

- **Phone Server IP:** Specifies your user name and the IP address of your iQ6 or Telos VX phone server. Login information is in the format: **user:pass@ip**. For example, if your VX user name is Marty and your password is "C185", you would enter **Marty:C185@192.168.0.123** where 192.168.0.123 is the IP address of your phone server.
- **Studio Name:** This is your studio name, as configured in the VX system.
- **Show Name:** This can be the same as your Radius show profile for consistency but it must correspond to a **Show Name** configured in your VX phone sys-

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tem. If you leave this blank, the current VX show will remain in use.

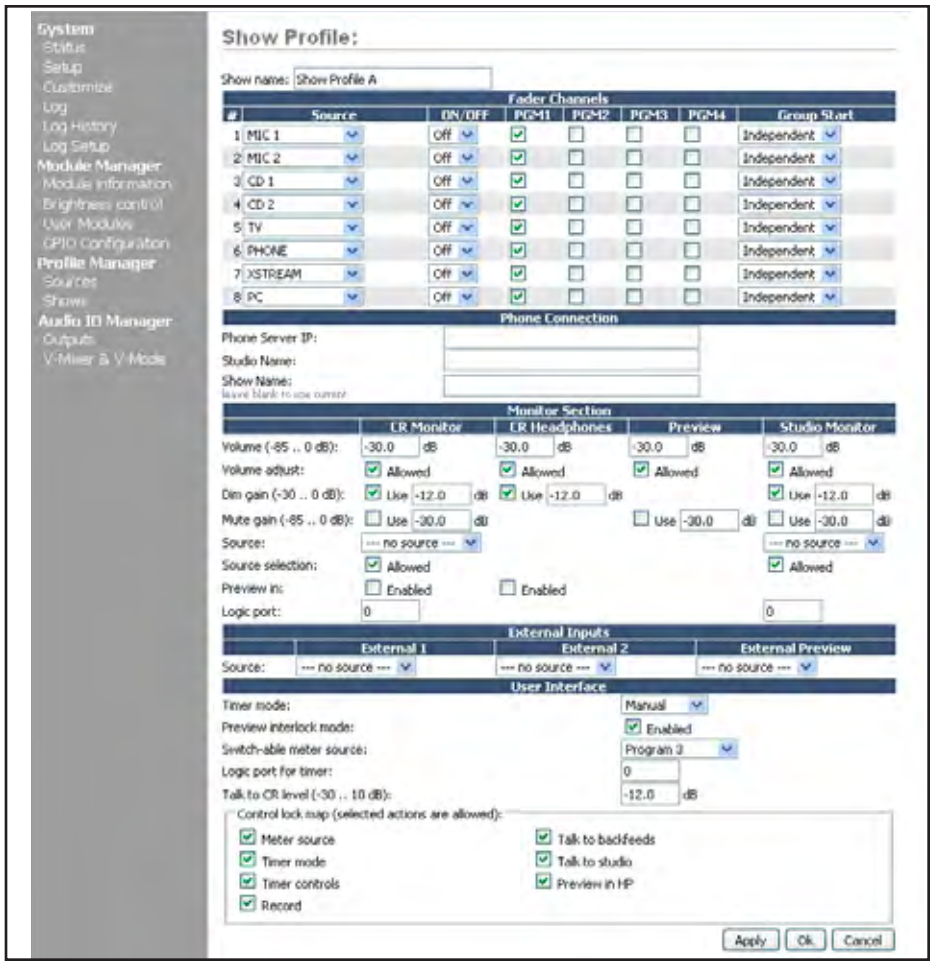
System
Status
Setup
Customize
Log
Log History
Log Setup
Module Manager
Module Information
Brightness control
User Modules
GPIO Configuration
Profile Manager
Sources
Shows
Audio IO Manager
Outputs
V-Mixer & V-Mode
Show Profile:
Show name: Show Profile A
Fader Channels
Source ON/OFF PGM1 PGM2 PGM3 PGM4 Group Start
1 MIC 1 Off Independent
2 MIC 2 Off Independent
3 CD 1 Off Independent
4 CD 2 Off Independent
5 TV Off Independent
6 PHONE Off Independent
7 XSTREAM Off Independent
8 PC Off Independent
Phone Connection
Phone Server IP:
Studio Name:
Show Name:
Monitor Section
CR Monitor CR Headphones Preview Studio Monitor
Volume (-85 .. 0 dB): -30.0 dB -30.0 dB -30.0 dB -30.0 dB
Volume adjust: Allowed Allowed Allowed Allowed
Dim gain (-30 .. 0 dB): Use -12.0 dB Use -12.0 dB Use -12.0 dB
Mute gain (-85 .. 0 dB): Use -30.0 dB Use -30.0 dB Use -30.0 dB
Source: --- no source --- --- no source ---
Source selection: Allowed Allowed
Preview in: Enabled Enabled
Logic port: 0 0
External Inputs
External 1 External 2 External Preview
Source: --- no source --- --- no source --- --- no source ---
User Interface
Timer mode: Manual
Preview interlock mode: Enabled
Switch-able meter source: Program 3
Logic port for timer: 0
Talk to CR level (-30 .. 10 dB): -12.0 dB
Control lock map (selected actions are allowed):
Meter source
Timer mode
Timer controls
Record
Talk to backfeeds
Talk to studio
Preview in HP
Apply Ok Cancel

Figure 4-2: Radius Show Profile Options

Monitor Section

- **Volume (-85..0 dB):** You may specify initial volume settings for the outputs listed below. The default setting is full mute:
  - » CR Monitor
  - » CR Headphones
  - » Preview
  - » Studio Monitor
- **Volume Adjust:** This checkbox determines whether you will permit the operator to change the volume settings for the outputs listed below — useful for locking a console into a certain state during unattended operation. "Allowed" is checked by default.
  - » CR Monitor
  - » CR Headphones
  - » Preview
  - » Studio Monitor
- **Dim Gain (-30..0 dB):** You may adjust the Dim Gain independently for CR Monitor, CR Headphones and Studio Monitor. This determines the amount of dimming that will be applied when Talkback is directed to one of these outputs. The default setting is -12.0 dB.
- **Mute Gain (-85..0 dB):** You can specify the amount of attenuation that takes place during a Mute function. The default is full mute. Mute can be applied to CR monitor, Preview and Studio Monitor but it is not applied to CR Headphones.
- **Source:** Selects the default source for CR Monitor and Studio Monitor. Available options are Program 1/2/3/4 and External 1/2.
- **Source Selection:** You may prevent the operator or studio talent from changing their CR Monitor or Studio Monitor source if the corresponding "Allowed" check box is unchecked.
- **Preview In:** determines whether or not Preview will be heard in CR Monitor or CR Headphones in addition to the Preview speaker.
- **Logic Port:** Use this to specify the Livewire channel number assigned to logic ports (GPIO) used for CR Monitor and Studio Monitor functions. These GPIO ports are used for on-air lights and other functions as described in the GPIO tables. Please refer to **Chapter 5: Configuring GPIO** for more details.

External Inputs

- **Source:** Lets you specify two External sources for monitoring purposes plus one source that is an External Preview source. Note that the drop-down selector displays all sources that have been defined as

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Radius sources in the Source Profile manager.

What's External Preview for? "External" refers here to any source not generated inside the studio you're working in. Let's say you have a Talkback source that you want to feed to the console's Preview speaker. To do this, you would setup a source profile for the Talkback audio and specify that source as External Preview

User Interface

Several global settings are defined in this section. Remember that even though these are general settings, they are specific to each Show Profile so you may accommodate the unique requirements of different shows.

- **Timer Mode:** is used to specify the default timer mode - manual or automatic. In automatic, the timer may be started by turning a channel ON.
- **Preview Interlock Mode:** If **Enable** is checked, pressing any fader's Preview key causes any other, previously-lit Preview key to be extinguished, removing that source from Preview**.** The board operator can assign multiple faders to Preview by pushing and holding any lit Preview key while pushing other Preview keys to add or subtract. If unchecked, each Preview key works like a press-on, press-off latching switch. Multiple faders can be assigned to Preview simply by pushing once; a second push cancels any individual Preview assignment.
- **Switchable Meter Source:** The third overbridge meter is a "soft" meter. The desired default may be selected from the drop-down menu. Choices are: **Program 3, Program 4** or **Follow Monitor.**
- **Logic Port for Timer:** Enter the Channel Number of the GPIO port you wish to use for external control of your Count-Up timer.
- **Talk to CR Level: Use** up to -30 dB attenuation or 10 dB gain on the contents of the Talkback bus.
- **Control Lock Map:** This section includes several check boxes. When a box is checked, the selected actions are allowed. Any unchecked boxes will prohibit the operator from accessing these functions when this Show Profile is loaded. The selections are:
  - » Meter Source
  - » Timer Mode
  - » Timer Controls
  - » Record Mode
  - » Talk to backfeeds
  - » Talk to Studio
  - » Preview in HP

As usual, please remember to hit **Apply** when you are done to save your changes.

What's Next?

We have mentioned GPIO many times in the preceding chapters. Join us in Chapter 5 to learn how to interface the logic functions of your Radius console with the real world.

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Chapter Five:

Configuring GPIO

Since the days of rotary-switch source selectors, it's been a challenge to properly route machine logic along with audio. Some complicated schemes have been devised over the years, but broadcasters wanted a simple, fast way to associate logic with audio in a routable environment.

Since the Axia IP-Audio system was designed as a true computer network, we were able to start with a clean sheet of paper to design the first truly routable machine logic/audio interface. Unlike conventional logic connections which require each command circuit to be wired individually, Axia sends machine controls over the same Ethernet your audio travels on, further reducing infrastructure, cost and tedium.

Along with controlling external audio devices, Radius' GPIO (General Purpose Input/Output) interface also includes logic commands for routine studio/control room operations such as tally lights, monitor muting, On-Air lights and more.

Chapter 3 explained source profiles and showed you how you can associate GPIO with a source. This chapter explains how to set up Radius GPIO to handle these functions. We also invite you to read the **GPIO Node User's Manual**, for more in-depth information on the topic of GPIO.

> ***Note:*** The Radius control surface is used with a Radius Core that includes four GPIO ports. GPIO ports on standalone GPIO nodes or other Axia consoles may also be associated with your Radius sources if your Radius is a part of a larger network. See **Chapter 7: Livewire Networks and Radius** for details.

GPIO Port Definitions

The Radius Core has four DB-15 connectors on its back panel. Each connector (also known as a GPIO port) can be associated with a device in your studio, and provides five opto-isolated inputs and five opto-isolated outputs per device for machine control, lamp drives and remote channel controls. GPIO ports can also be monitored and controlled by our PathfinderPC software.

GPIO ports are programmed to support several different types of devices. How does a GPIO port "know" which type of device is assigned to it?

Back in **Chapter 2**, when you constructed a Source Profile for a telephone hybrid, you defined the source type (see "Adding Backfeeds and GPIO to a Source Profile"). This is important, because when that source is assigned to a console fader, Radius uses this Source Profile selection to tell the GPIO Node what sort of command to send to the attached device.

If Radius "sees" in the Source Profile that the assigned device is a microphone, it tells the GPIO Node to send logic for **On, Off, Remote Mute** and Remote **Talk** commands on the appropriate pins. If it "sees" a line input, it tells the GPIO Node to send **Start, Stop** and **Reset** commands, plus closures for **Ready** lights, etc.

Axia GPIO ports can deliver unique command sets for the following types of devices:

1. Mic (Operator, Guest, Producer)
2. Line Input
3. Codec
4. Telephone Hybrid
5. Computer Playback Device
6. Control Room Monitor
7. Studio Monitor
8. Profanity Delay Device
9. Recording Device
10. Accessory Button Panel Device

The next few pages contain tables that explain what function the pins provide in each different device mode.

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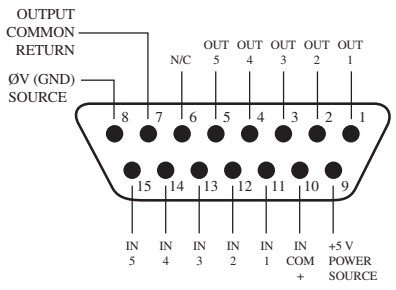
GPIO Operator's Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to Monitor 2) Command	13	Active Low Input	Activates the TALK TO MON2 function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to PREVIEWED SOURCE) Command	15	Active Low Input	Activates the TALK button on every source currently in preview and routes mic audio to the Talkback bus.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to Monitor 2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK TO MON2 is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to PREVIEWED SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PREVIEWED SOURCE is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic +5 Volt Supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Input Common	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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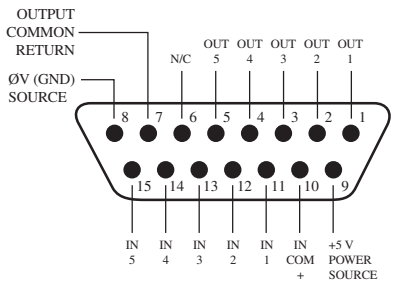
GPIO Control Room Guest Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
NOT CONNECTED	15		
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
NOT CONNECTED	5		
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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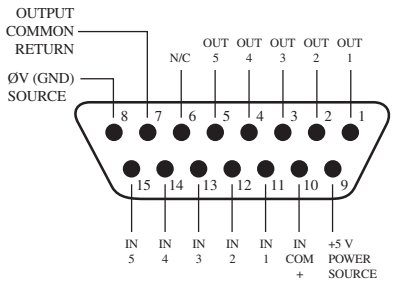
GPIO Studio (Monitor 2) Guest Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
NOT CONNECTED	15		
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
NOT CONNECTED	5		
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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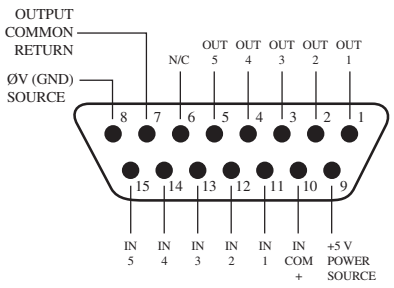
GPIO Producer's Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to MONITOR 2) Command	13	Active Low Input	Activates the TALK to MON2 function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to PREVIEWED SOURCE) Command	15	Active Low Input	Activates the TALK button on every source currently in Preview and routes mic audio to the Talkback bus.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to MONITOR 2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK to MON2 is active.
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to PREVIEWED SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PREVIEWED SOURCE is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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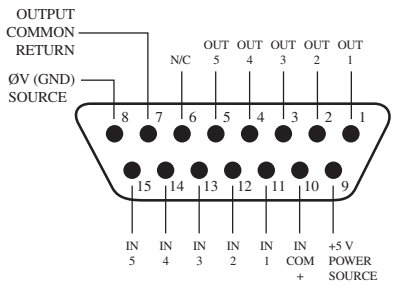
GPIO Line Input Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PREVIEW Command	13	Active Low Input	Turns preview ON
RESET Command	14	Active Low Input	Turns channel OFF, while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from OFF to ON
STOP Pulse	5	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from ON to OFF
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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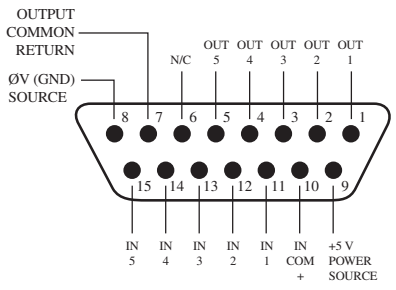
GPIO Codec Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external button to activate channel TALK TO SOURCE function.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE are active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF.
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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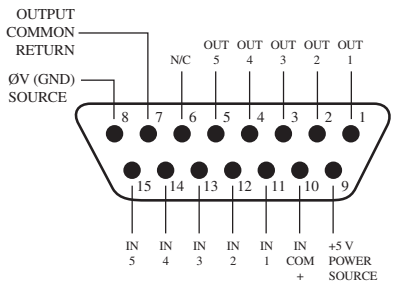
GPIO Telephone Hybrid Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
PREVIEW Command	13	Active Low Input	Turns preview ON
RESET Command	14	Active Low Input	Turns channel off while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 ms PULSE is sent when channel is first turned ON or when PVW is first selected
STOP Pulse	5	Open Collector to Logic Common Return	A 100 ms PULSE sent when channel is turned OFF.
POWER & COMMAND			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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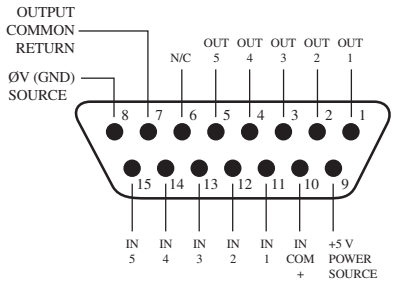
GPIO Control Room Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE CR Command	11	Active Low Input	Mutes CR monitors and Preview speakers
DIM CR Command	12	Active Low Input	Allows external dimming of CR monitor speakers.
Enable EXT PREVIEW Command	13	Active Low Input	Feeds External Audio Input to PREVIEW
TALK TO EXT Command	14	Active Low Input	Turns on Talk to External Audio.
Not used.	15	Active Low Input	
OUTPUTS			
CR ON AIR Lamp	1	Open Collector to Logic Common Return	Illuminates whenever CR monitors are muted
DIM CR Lamp	2	Open Collector to Logic Common Return	Illuminates whenever control room monitors are DIMMED
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is active.
TALK TO EXT Lamp	4	Open Collector to Logic Common Return	Illuminates when Talk to External is active.
TALK (to CR) Active Lamp	5	Open Collector to Logic Common Return	Active whenever a source has activated its TALK (to CR) function
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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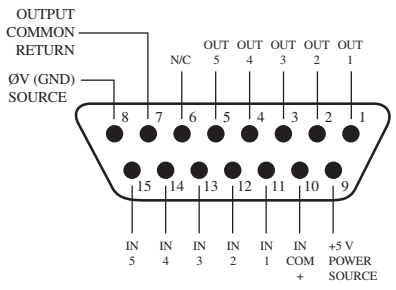
GPIO Studio Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE Studio Command	11	Active Low Input	Allows external muting of Studio monitor speakers.
DIM Studio/Monitor 2 Command	12	Active Low Input	Allows external dimming of Studio monitor speakers.
Remote Event Timer Trigger Command	13	Active Low Input	Resets Count-up Event Timer to zero and starts timer.
NOT CONNECTED	14		
Not Used	15	Active Low Input	
OUTPUTS			
Studio ON AIR Lamp	1	Open Collector to Logic Common Return	Illuminates whenever Studio monitors are muted
DIM Studio/Monitor 2 Lamp	2	Open Collector to Logic Common Return	Illuminates whenever Studio monitors are DIMMED
Event Timer Trigger Output	3	Open Collector to Logic Common Return	A 100 ms PULSE sent when event timer is started from zero.
NOT CONNECTED	4		
TALK TO STUDIO/MONITOR 2 Active Lamp	5	Open Collector to Logic Common Return	Illuminates whenever the TALK TO STUDIO/MON2 function is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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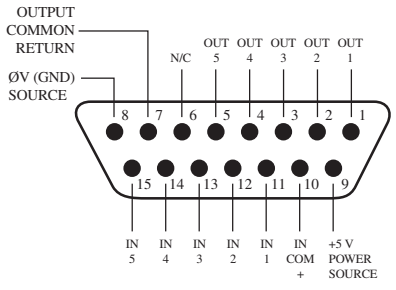
GPIO Computer Playback Device Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PREVIEW Command	13	Active Low Input	Turns preview ON
Not Used	14	Active Low Input	
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
NEXT Pulse	1	Open Collector to Logic Common Return	A 100 mS PULSE sent when ON button is depressed, except when initially turned ON.
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 ms PULSE sent when channel is first turned ON.
STOP Pulse	5	Open Collector to Logic Common Return	A 100 ms PULSE sent when channel is turned OFF.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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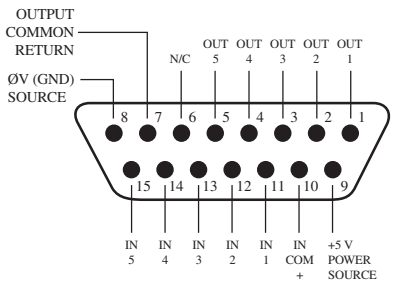
GPIO Film Legendable User Button Module / Accessory Panel / Rack Panel

Name	Pin	Type	Notes
INPUTS			
User Button 1 Lamp	11	Active Low Input	Illuminates User Button 1 Lamp
User Button 2 Lamp	12	Active Low Input	Illuminates User Button 2 Lamp
User Button 3 Lamp	13	Active Low Input	Illuminates User Button 3 Lamp
User Button 4 Lamp	14	Active Low Input	Illuminates User Button 4 Lamp
User Button 5 Lamp	15	Active Low Input	Illuminates User Button 5 Lamp
OUTPUTS			
User Button 1 Switch	1	Open Collector to Logic Common Return	Active when User Button 1 is pushed
User Button 2 Switch	2	Open Collector to Logic Common Return	Active when User Button 2 is pushed
User Button 3 Switch	3	Open Collector to Logic Common Return	Active when User Button 3 is pushed
User Button 4 Switch	4	Open Collector to Logic Common Return	Active when User Button 4 is pushed
User Button5 Switch	5	Open Collector to Logic Common Return	Active when User Button 5 is pushed
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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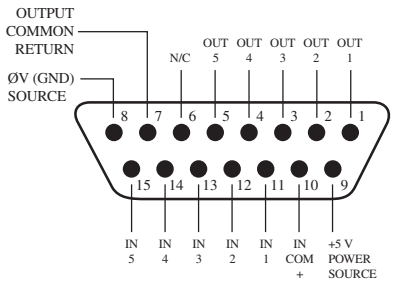
GPIO Recording Device Logic

Name	Pin	Type	Notes
INPUTS			
Not Used	11	Active Low Input	
Not Used	12	Active Low Input	
Not Used	13	Active Low Input	
Not Used	14	Active Low Input	
Not Used	15	Active Low Input	
OUTPUTS			
Rec ON Lamp	1	Open Collector to Logic Common Return	Active whenever Record Mode is engaged.
Rec OFF Lamp	2	Open Collector to Logic Common Return	Active whenever Record Mode is NOT engaged.
Not used	3	Open Collector to Logic Common Return	n/a
START Pulse	4	Open Collector to Logic Common Return	A 100 mS PULSE sent when Record Mode is engaged.
STOP Pulse	5	Open Collector to Logic Common Return	A 100 mS PULSE sent when Record Mode is disengaged.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



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Assigning GPIO to a Source

As you've seen by studying the previous pages, a lot of the work of assigning logic to a source is done for you; once a GPIO port is linked with a Source Profile, all that remains to do is to solder cables connecting the GPIO's DB-15 connectors to the device's control interface.

So, how *do* you link a GPIO port with a Source Profile? It's very easy; let's do it step-by-step.

> ***Note:*** This procedure assumes that you have already defined at least one Source Profile using the instructions outlined in Chapter 2.

1. Open your Web browser and enter the IP address of your Radius console. Choose **GPIO Configuration** from menu. Enter your password if prompted (default login is "user", leave the password field blank).

2. If you haven't previously assigned any GPIO ports, the GPIO definitions screen will be blank. Notice the status indicators at the top of the page, showing the state of the input and output pins of each port. Click on the list icon to the right of an unused line.

3. When you click on any list icon, a small popup window will open, enumerating all of the audio sources available on the Livewire network (Figure 5-2). Choose the source you wish to associate with a GPIO port by clicking on it; the window will close and the source's name and channel number will appear in the **Channel** box.

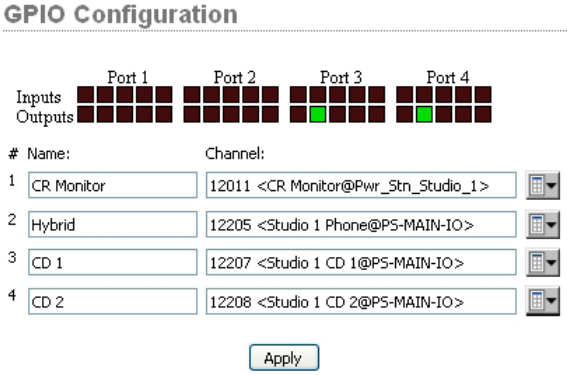

Figure 5-1: GPIO Port Assignment

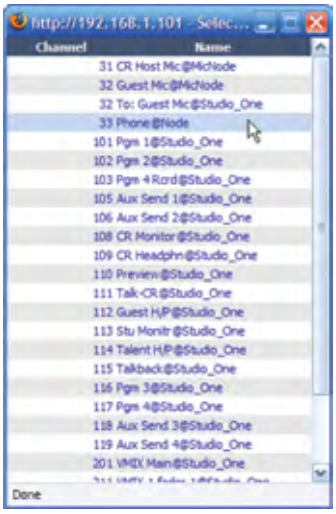

Figure 5-2: GPIO Select Source popup list

4. Type a descriptive name in the **Name** field, and click on the **Save** button at the bottom of the page.

5. Look at the page-top status indicator for Port 1. You'll see that one of the pin status indicators is lit in green; this means that the port is now sending a GPIO logic state of "true" through this pin. Assign the source for which you just created a GPIO link to an Radius channel; operate the **On** and **Off** keys for the channel and watch the pin status indicators change as you do so.

The source we've been using for this demonstration is a telephone hybrid; we can now observe the pin status indicators change as we turn the channel on and off, as shown in Figure 5-3.

Referring to the Hybrid Logic Chart, we can see that when the Radius channel is **Off**, the indicator representing Output Pin 2 – the **Off Lamp** logic command – is high. Turning the channel **On**, we see Pin 4 pulse

Figure 5-3: Pin status indicators showing GPIO port activity

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briefly before Pin 1 goes high: the GPIO has just sent a **Start** pulse, then lit the **On** Lamp. If you turn the channel **Off** again, you'll observe a **Stop** pulse, and the **Off Lamp** command once again become active.

See how easy that was? Simply assigning an existing audio source to a GPIO port automagically configures the port for the type of device supplying the audio, and send the appropriate logic commands to that port when the source is assigned to a Radius channel.

Connections to GPIO Ports

Input Connections

Current on these inputs must be limited to 20 mA or less, through the use of a current-limiting resistor. An external power source (24 volts DC maximum) is recommended for all inputs and outputs, to prevent ground loops between equipment. However, if customer equipment is completely isolated, using power from the GPIO port connectors is acceptable. Figure 5-4 shows details for both types of connections:

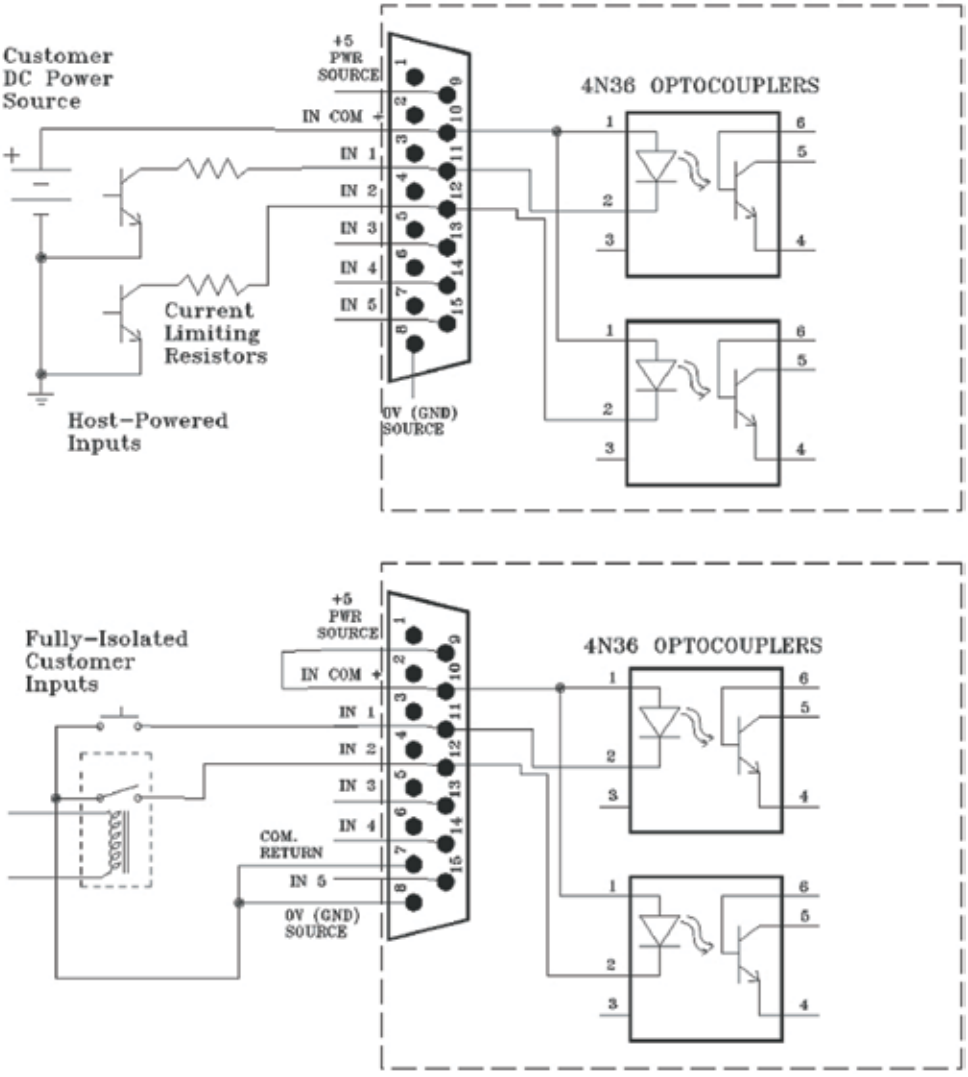

Figure 5-4 GPIO input connections

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Output Connections

The GPIO port's outputs are opto-isolated. Current should be limited 100 mA through each output, with the total current draw from the +5 Volt supply not to exceed 3 amps. Figure 5-5 shows the recommended connections for outputs:

NOTE: Some external devices will allow a logic control input "sink to ground" to activate. Thus, some devices may not work with an Axia GPO control, because they may not fully achieve ground through the output transistor. The voltage drop between the collector and emitter may not be low enough to activate the device, so an external relay controlled by the GPO may need to be used to provide a "dry" contact closure to the external device.

Please note that this section is provided as a "jump-start" introduction to Axia GPIO nodes. For a fuller understanding of the GPIO node's options and requirements, you may wish to read the *GPIO Node User's Manual*.

What's Next

When you're ready, join us in Chapter 6 for a comprehensive "under the hood" discussion of Radius Advanced Controls.

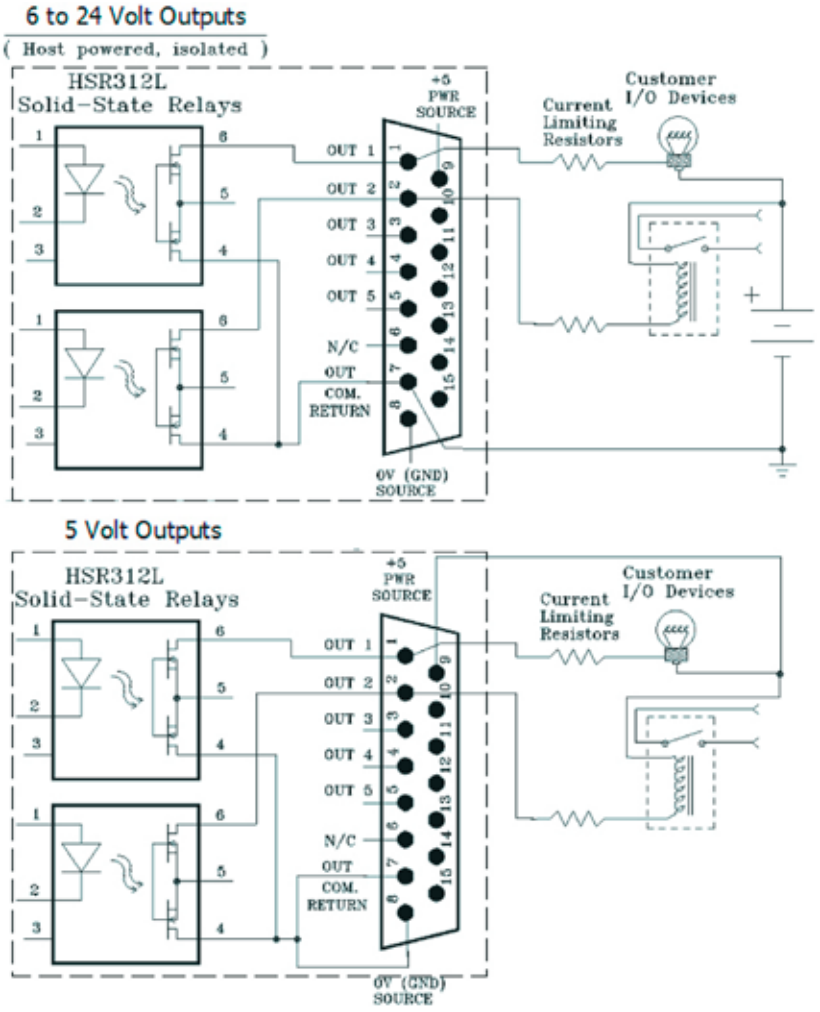

Figure 5-5: GPIO output connections

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Chapter Six:

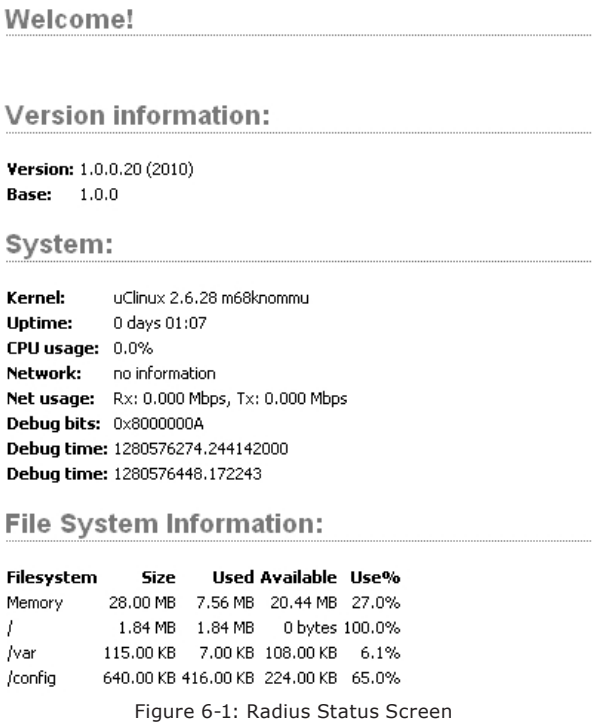
Advanced Controls

In previous chapters, we covered the basics of setting up your Radius. Now that you're familiar with your console, you should consider yourself an advanced user! In keeping with your dawning guru status, it's time for a thorough look at some of Radius's more advanced configuration options.

Many of Radius's features can be customized by software settings. And owners like you have given us great ideas for new operational capabilities, some of which are included in our periodic software updates. With that in mind, we strive to keep our documentation as up-to-date as possible.

If you have questions that are not covered here, we invite you to contact our 24/7 Support crew:

- by email at support@AxiaAudio.com
- by phone at +1 216-241-7225.


Welcome!
Version information:
Version: 1.0.0.20 (2010)
Base: 1.0.0
System:
Kernel: uClinux 2.6.28 m68knommu
Uptime: 0 days 01:07
CPU usage: 0.0%
Network: no information
Net usage: Rx: 0.000 Mbps, Tx: 0.000 Mbps
Debug bits: 0x8000000A
Debug time: 1280576274.244142000
Debug time: 1280576448.172243
File System Information:
Filesystem Size Used Available Use%
Memory 28.00 MB 7.56 MB 20.44 MB 27.0%
/ 1.84 MB 1.84 MB 0 bytes 100.0%
/var 115.00 KB 7.00 KB 108.00 KB 6.1%
/config 640.00 KB 416.00 KB 224.00 KB 65.0%
Figure 6-1: Radius Status Screen

Radius Configuration Screens

The Radius Control Center

Figure 6-1 shows the screen that greets you when you log into your Radius from your Web browser. The main menu runs down the left side and gives access to options and settings. In the main window you can see at a glance what version of Radius software is loaded, system information, CPU statistics, and more. For diagnostic purposes, this information screen is openly accessible, but clicking on any menu heading will prompt you for a password. Here you'll find options related to the Radius Core and Ethernet switch.

The Status Screen

The Status screen is your home page. It displays basic system status as well as information pertaining to the active software load. A sample of this screen is shown in figure 6-1. Additional information displayed may include::

- Radius software version info including the current active Version and Base
- System Information including:
  - » Version of Linux kernel
  - » Uptime of the Radius expressed in days, hours and minutes
  - » CPU utilization
  - » Network Information
  - » Network status and utilization - transmit and receive data rates expressed in Mbps.
  - » Debug information
- File System Information - provides information on memory and solid state disk utilization for the operating system. This information may be requested by Axia tech support.

The Setup Menu

We touched briefly on the IP features of the Setup screen in Chapters 1 and 2. Now let's look at the rest of the tools on this page shown in Figure 6-2.

- **Network Configuration** contains the fields for the Radius IP Address and other network functions.
  - » **Hostname** is the unique name you give the console to identify it in the Web interface. You might

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want to name it after the studio it's in.

» **IP Address:** This is the network address of your Radius. We suggest that you have a plan to ensure that all of your Livewire devices are assigned unique, IP addresses.

Shortcut Key: You can also set the IP address of your Radius by simultaneously pressing and holding the **Meter** and **Clock** keys on the Radius Console. Use the Studio Monitor knob to navigate and follow the prompts to give your Radius a new IP address.

» **Netmask:** Enter your network's submask here. If you are not sure about this - consult your network administrator. In most cases a Netmask of 255.255.255.0 is suitable.

» **Gateway:** If you have a Core switch or a network gateway to allow access from outside your facility, enter that address here. A gateway is required for advanced IP address configurations and for connection to an external NTP server.

Note: A reboot is required when any changes are made to the network parameters on the Radius Setup Menu.

- **Web Access Password:** By default, the user name is set to "user" and the password is blank. If you want to add a password, type it in both boxes and click **Change.** Note that passwords will not display here once changed, so be sure to make a record of all password changes.

Figure 6-1: Radius Setup Screen

- **Software Update:** This is where you upload new Radius operating software.

» **Software Versions** shows you the software packages currently stored on your Radius flash memory card. The blue radio button is displayed next to the active software image.

» **Change Active Bank:** When you want to change the active software package, click on the radio button next to the image you wish to activate, and click this button. A reboot is necessary to load the new software choice.

» **Upload Image:** From time to time, Axia makes new software packages available for download from AxiaAudio.com/downloads/. Use the **Browse** and **Upload Image** buttons to install new software, as described in the Technical Bulletin that accompanies each software release. Note that uploading a new image will always over-write the inactive bank.

» **Reboot** shuts down Radius and restarts it. A reboot is required after changing active software or after changing any settings in your Network Configuration.

Save your work! We recommend backing up your Configuration files whenever a major change is made, or is about to be made, to your Radius. Whether you are re-configuring Show or Source profiles *en masse* or about to perform a software update, it's always a good idea to have a copy of your important data.

The Customize Menu

The **Customize** page contains Clock settings and Backup/Restore functions. Future enhancements may include additional features in this area.

Clock

» **Clock Format:** This is where you define whether you will view your time in 12 hour or 24 hour format. Each individual Radius in your network can use either its own internal clock or an external time server for timekeeping. The next group of set-

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tings will set up NTP (Network Time Protocol) time sync.

» **Time Zone Offset:** Specify your local GMT offset in hours and minutes. For example, if you live in the Eastern Tome zone in the USA, your GMT offset will be -5 hours, 0 minutes. If you live in Riga, Latvia, your offset will be +2 hours.

» **DST in effect (+1 hour):** In those area that implement Daylight Savings Time, this check box is used to tell your system to add one hour to the GMT offset.

» **NTP:** The check box should be checked if your Radius's NTP clock sync function is enabled. To disable NTP service, simply un-check this box. The other box here is used to enter the IP address of your external NTP server. This server may be a PC on your local network or it may be the IP address of a public NTP server in the Internet. Note that you must specify an IP address – a URL will not work.

If you wish to connect to a public or external NTP server, be sure to specify a valid Gateway in your IP configuration.

You can tell if your NTP connection is working by observing the presence of the NTP indicator next to the clock display on your Radius overbridge.

Configuration

This is an extremely important area of your Radius configuration since this is where you will backup all of the hard work you have done.

» **Backup Settings:** All Radius settings can be saved in a single operation and backed up to a location on your local drive. When you select the **Backup** option, you will see the familiar location navigator that prompts you for the folder in which you wish to store your backup file. The backup file will be named according to the device name, IP address and date/time the backup was performed. This naming conventions allows you to store multiple backups from multiple devices in the same location. An example of a backup file name might be **Radius-192.168.0.200-2010-03-25_17-13.xml**.

» **Restore Settings:** To restore a previous Radius backup, use the browse button to locate the backup and then select **Restore.**

Log, Log History and Log Setup Menus

Radius keeps a detailed log of user actions and other system events. Choosing the **Log** menu item lets you examine the last 100 logged activities; choosing the **Log History** menu item allows you to view an entire day's log at one time and delete old log files if you choose. The oldest log files are deleted automatically as storage space dictates.

The **Log Setup** menu item allows you to specify an external Syslog server and to configure the debug level. This menu also allows you to specify from eight different severity level filters. The filter choices (in order of decreasing severity) are:

- Emergency: system us unusable
- Alert: action must be taken immediately
- Critical: critical condition
- Error: error conditions
- Warning: warning conditions
- Notice: normal but significant condition
- Informational: informational messages
- Debug: debug level messages

Use this section as directed to do so by Tech Support.

The Module Manager

The **Module Manager** heading displays information about the Radius control surface as well as accessories connected to your console. Some of these fields are for diagnostics; below is a list of those you might find useful:

» **Address:** displays the number Axia has assigned to each connected unit.

» **Type:** gives you the working name of the module.

» **State:** tells you whether or not the module is working, i.e., communicating with the Core CPU.

» **Ping**: Indicated ping response in ms.

» **Messages, Idle Time, Uptime:** All indicate diagnostic information that is related to the CAN-Bus communications. Tech support may request

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this information for troubleshooting purposes.

» **Hardware Revision:** displays the hardware version of the Radius module.

» **Firmware Version:** displays the firmware version that's loaded on the Radius module.

» **Upload Firmware Group:** From time to time, new module firmware may become available from Axia. You may install new firmware by choosing Browse and then Upload.

NOTE: Updating a module takes it offline during the update process. While audio playing through the module's faders will continue uninterrupted, keep in mind that while updating (usually about 10-15 seconds) you will not be able to adjust gain or turn faders on or off on the module being updated.

Brightness Control Menu

This menu offers several options for fine-tuning your Radius displays to your studio's ambient lighting. All values are expressed in percent. The default setting will be suitable for most studio environments, however you may make adjustments as needed.

User Modules Menu

This menu allows you to set up and use Axia accesso-button modules and accessory panels with your Radius console. Your accessory panels are supplied with a Users Manual that explains their setup and operation.

GPIO Configuration Menu

This menu is where you configure the assignments for the four opto-isolated GPIO connections found on the rear panel of the Radius Core. For a thorough discussion of this screen and how to configure GPIO, please refer to **Chapter 5: Configuring GPIO**.

Profile Manager Menu

These menus allow you to construct and administer Show and Source Profiles specific to this Radius console. Refer to **Chapter 2: Inputs and Outputs** and **Chapter 4: Show Profiles** for details on Source and Show profiles.

IO Manager Menu

These menus allow you to administer your Radius's audio Inputs and Outputs. Refer to **Chapter 2: Inputs and Outputs** for complete details on configuration of these controls.

Ethernet Switch settings (Preconfigured)

Radius Core contains an integrated Ethernet switch. This switch manages the multicast functions of the console and enables it to be connected to other devices via your managed Livewire LAN.

NOTE: We make every effort to keep our manuals up-to-date, but sometimes R&D works quicker than our documentation elves! Controls documented in this section may vary from those actually presented on your product. Please contact support@AxiaAudio.com for explanations of any differences you may find.

What's Next

Radius is designed to be the perfect standalone IP console. But if you want to make it a part of a larger Livewire studio network — no problem! Read on as we discuss networking options for your Radius.

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Chapter Seven:

Networking with Radius

Radius is a superb standalone broadcast console — but like all Axia broadcast products, Radius has networking capabilities as well.

Axia invented networked audio for broadcast. We call it Livewire, and Radius uses this networking ability in several different ways:

- Your Radius console can load and utilize sources from other studios connected to your Livewire network.
- You may deliver outputs from your Radius to the Livewire network for use by others.
- Naturally, Radius can also use GPIO logic that is available on your Livewire network.
- The Livewire network allows you to use our companion software package, PathfinderPC, for numerous advanced functions. The list of features provided by PathfinderPC is growing continuously.
- Up to four Radius consoles can be configured daisy-chain style, without the need for an external core Ethernet switch, to take advantage of network redundancy provided by implementation of the Spanning Tree Protocol (STP) standard.
- You can manage your Radius remotely using its web browser interface.
- Configuration backup and restore operations may be performed over the network as well.

For an in-depth discussion of Livewire, please review our publication "**Introduction to Livewire: IP-Audio System Design Reference and Primer,**" which provides an excellent discussion of IP-Audio and how standard Ethernet can be used to transport real-time broadcast audio in a studio environment. This is available at AxiaAudio.com/manuals/.

Inputs

Your Radius console has a custom, built-for-broadcast Ethernet switch built right in.

When properly configured and connected to other Radius consoles (or, in a large studio facility, to an Ethernet "core" switch), this network connection permits you to access the resources of other devices connected to your network.

Radius has a 12-in, 4-out "gateway" for exchanging audio data with the rest of your Livewire network. Radius can consume up to 12 Livewire audio streams *from* the network, and it can deliver 4 Livewire audio streams *to* the network. To understand this, it might help to envision this network connection as a high-speed freeway, with 12 lanes coming towards your studio and 4 lanes going out.

Selecting the Source Input

In **Chapter 2: Inputs and Outputs**, we discussed the Radius console's local inputs - those associated with the XLR and RJ-45 connectors on the rear panel of the Radius Core. We saw how we could associate each of these physical inputs with a Source Profile. The type of source (CR mic, Guest mic, codec, phone, etc) determined parameters such as monitor muting and backfeeds.

When each Source Profile is configured, you have a choice for the Source Input. If you select one of the rear panel connectors, the console source is associated with that input. If you select **Livewire**, you are telling the Radius that this source is not associated with a local input but is, instead, a source from elsewhere on your Livewire network. You may then assign that **Primary Source** by using the Browse button to choose from a list of Livewire sources. That's all there is to it!

Figure 7-1 shows an example of a codec source that's been configured as a Livewire Source Input with a network Primary Source. In this example, the Livewire **Channel Number** of that source is 1037 and it is coming from an Axia node named AES-103. The source has a designation of CODEC_1 on that node. Everything else

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Source Profile:
Source type: Codec
Source name: Codec 1
Source input: Livewire
Primary source: 1037 <CODEC_1@AES-103>
Signal mode: Stereo
Record mode: Stereo
Fader trim gain (-25 .. 25 dB): +0.0 dB
Panorama position: (-24 .. 24): 0
Source availability:
Channel 1 Channel 2 Channel 3 Channel 4 Channel 5 Channel 6 Channel 7
Channel 8 Channel 9 Channel 10 Channel 11 Channel 12 Channel 13 Channel 14
Channel 15 Channel 16 Channel 17 Channel 18 Channel 19 Channel 20 Channel 21
Channel 22 Channel 23 Channel 24 External V-Mixer
Fader mode: Normal
Preview mode: Pre-Fader (CUE)
Preview switching:
Channel ON turns Preview OFF
Preview ON turns Channel OFF
Auto-start timer: Enabled
Logic port: Exclusive mode
Feed to source: Auto (Program1 / Phone)
Feed to source dim gain (-30 .. 0 dB): Enable -10.0 dB
Live Controls:
SOURCE Disabled Enabled Primary
GAIN Disabled Enabled Primary
MODE Disabled Enabled Primary
PANORAMA Disabled Enabled Primary
BACKFEED Disabled Enabled Primary
Save as Copy
Apply
Ok
Cancel

Figure 7-1: Codec Source

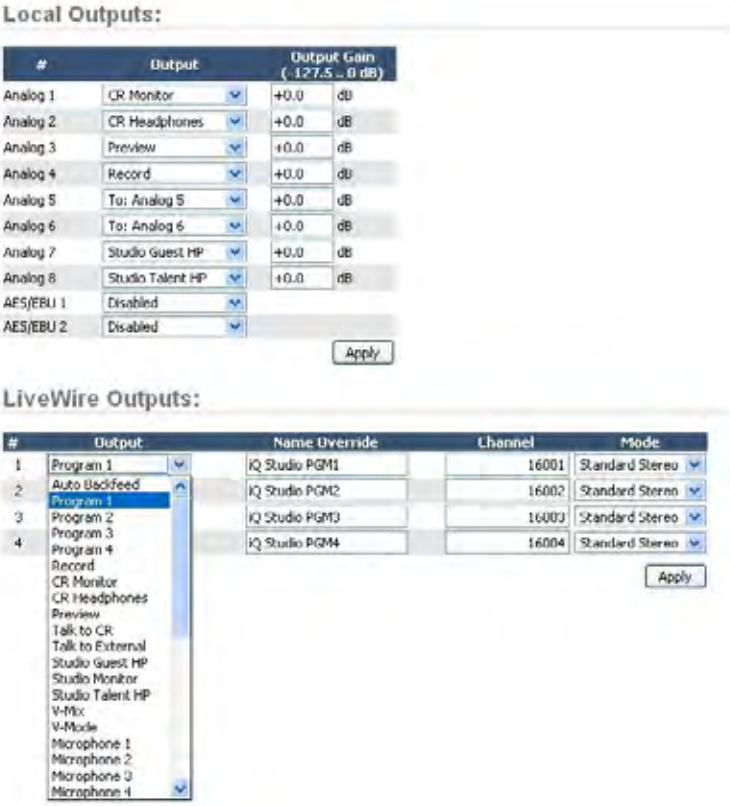

Figure 7-2: Livewire Outputs

in the configuration of the Source Profile is exactly the same as if that codec was connected directly to the Radius Core.

In summary, your Radius console sources inputs may be configured from any of the rear panel connectors or from as many as 12 incoming Livewire network sources for a total of up to 34 sources. These network sources may be redefined at any time, facilitating a highly flexible system!

Outputs

The Output configuration of your Radius has two sections: one for **Local Outputs** (rear panel connectors) and one for **Livewire Outputs**. As you might expect, the Local Outputs are usually configured for feeds to in-studio devices like CR Monitors, headphone amplifiers, etc.

The **Livewire Outputs** consist of four streams from your Radius that you wish to deliver to your Livewire network. These four streams will be available to other devices and users on your network.

The following parameters are configured for each of the four Livewire Outputs:

- **Output**: Specifies the audio feed to be delivered to the network on each of the four Livewire channels. Selections include:
  - » Any of the local inputs: 4 microphone, 16 stereo line or 2 digital AES inputs.
  - » Any internal generated Radius sources including PGM-1, 2, 3 or 4, Record, CR Monitor, CR Headphones, Preview, Talk to CR, Talk to External, Studio Guest HP, Studio Monitor, Studio Talent HP, and Auto Backfeed.
  - » VMix or VMode outputs.
- **Name Override**: Allows you to specify a name (16 characters maximum) that will

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describe this Radius source to network users. This field will override the source name specified in the local Source Profile.

- **Channel**: The Livewire channel designation. Specify a unique channel number between 1 and 32,767 that will be used by the Livewire network to identify your Radius's network audio.
- **Mode**: Enables and chooses the stream priority for this channel. Options are:
  - » **Disabled** - the channel is off.
  - » **Live Stereo -** Choose this when the audio output contains live voices or other high-priority audio.
  - » **Standard Stereo** - Choose when the audio output consists of pre-recorded music or programming.

GPIO

The Radius Core has four physical GPIO ports presented on DB-15 connectors. Often, these ports are associated with local sources for logic control of devices such as CD players or for external on/off control of your microphones. In these cases, select the local source to which you wish to map the GPIO port by using the **Logic For** drop down box.

Sometimes, however, you may wish to use GPIO logic that is associated with some other Livewire network audio source. In this case, you will select **Livewire** from the **Logic For** drop-down. You may then enter the Livewire channel number directly or by using the **Livewire** button on the right side of the screen. When you click on this button, a pop-up prompts you to select a Livewire channel from your network; the GPIO port will then become the logic interface for that Livewire source.

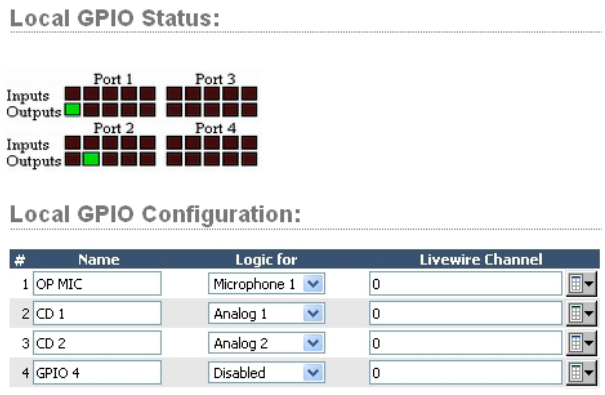

Figure 7-3: GPIO

Networking

To deploy Radius in a network setting, you will interconnect the Ethernet switch built into your Radius Core with other Radius Cores, or to external Ethernet switches. The custom Ethernet switch integrated into the Radius Core differs from an off-the-shelf network "core" switch in the following ways:

- The Ethernet switch in an Radius Core is always at the "edge" of a large Livewire network. In IT parlance, this type of switch is actually referred to as an "edge" switch.
- If you have a large Livewire LAN, you will usually have a central, high capacity, managed switch that handles all of the connections from your "edge" devices, including Radius Core switches. This central switch is referred to as your "core" switch. Common core switches include the Cisco 3750, 4948 and some of the 6400 series switches. The core switch is often completely populated with 1000BT ports.

IT Note: If your system employs an Ethernet core switch, the Ethernet core switch must have an IP address numerically lower than any of your Radius units or Ethernet edge switches. This function of the IGMP standard ensures that the multicast querier is handled by the core switch if one exists. The querier function will be passed off transparently to another switch should the core switch fail or be disconnected.

Work-in-progress Note: At the time of this writing, Networking details are being finalized. Please contact Axia Technical Support at support@AxiaAudio.com for assistance with networking settings.

We also recommend joining the Axia Tech mailing list to ensure that you receive immediate updates regarding new software or documentation releases. You can do so by clicking the "Sign Up Here" links at the top of the Axia Download and Manuals pages at AxiaAudio.com/manuals/ and AxiaAudio.com/downloads/.

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Mimicking nature,

streams combine to form anew

many become one.

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Appendix A:

Radius Specifications

Microphone Preamplifiers

Source Impedance: 150 ohms
Input Impedance: 4 k ohms minimum, balanced
Nominal Level Range: Adjustable, -75 dBu to -20 dBu
Input Headroom: >20 dB above nominal input
Output Level: +4 dBu, nominal

Analog Line Inputs

Input Impedance: 20 k Ohms
Nominal Level Range: Selectable, +4 dBu or -10dBv
Input Headroom: 20 dB above nominal input

Analog Line Outputs

Output Source Impedance: <50 ohms balanced
Output Load Impedance: 600 ohms, minimum
Nominal Output Level: +4 dBu
Maximum Output Level: +24 dBu

Digital Audio Inputs and Outputs

Reference Level: +4 dBu (-20 dB FSD)
Impedance: 110 Ohm, balanced (XLR)
Signal Format: AES-3 (AES/EBU)
AES-3 Input Compliance: 24-bit with selectable sample rate conversion,
20 kHz to 216kHz input sample rate capable.
AES-3 Output Compliance: 24-bit
Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
Internal Sampling Rate: 48 kHz
Output Sample Rate: 48 kHz
A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

Any input to any output: +0.5 / -0.5 dB, 20 Hz to 20 kHz

Dynamic Range

Analog Input to Analog Output: 102 dB referenced to 0 dBFS,
105 dB "A" weighted to 0 dBFS
Analog Input to Digital Output: 105 dB referenced to 0 dBFS
Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
Digital Input to Digital Output: 125 dB

Equivalent Input Noise

Microphone Preamp: -128 dBu, 150 ohm source, reference -50 dBu input level

Total Harmonic Distortion + Noise

Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input,
+18 dBu output
Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

Crosstalk Isolation, Stereo Separation and CMRR

Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kHz
Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz
Microphone Input CMRR: >55 dB, 20 Hz to 20 kHz

Power Supply AC Input, Radius Core and Console

Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
Power consumption: 300 Watts **(Preliminary Estimate)**

Operating Temperatures

-10 degrees C to +40 degrees C, <90% humidity, no condensation

Dimensions (HxWxD) and Weight

Radius Core 5.25 x 19 x 15 inches, 15 pounds (**Preliminary**)

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Table of Inputs and Outputs

Main Outputs	Type	Comments
Program 1	Stereo	Main stereo bus (Usually referred to as Program)
Program 2	Stereo	Second stereo bus (Sometimes referred to as Audition)
Program 3	Stereo	Third stereo bus (Sometimes referred to as Utility)
Program 4	Stereo	Fourth stereo bus (Sometimes referred to as Auxiliary)
Program 4 Record	Stereo	Fourth stereo bus variation (post fader, pre on/off)
Program 4 Phone (internal bus only)	Stereo	Fourth stereo bus variation (pre fader, pre on/off)
V-Mixer	Stereo	Stereo Submix - sum of one stereo source
V-Mode	Stereo	Used for processing MONO sources and creation of MONO streams

Monitor-related Outputs	Type	Comments
CR Monitor	Stereo	Control room monitor speakers, source and level controlled by Monitor 1 control
CR Headphone	Stereo	Control room (board op) headphone, source and level controlled by headphone control
Studio Monitor	Stereo	Studio monitor speakers, source and level controlled by Studio Monitor control
Studio Headphone Talent	Stereo	Studio (talent) headphones, source same as main, fixed level output, with talkback
Studio Headphone Guest	Stereo	Studio (guest) headphones, source same as main, fixed level output, no talkback
Preview	Stereo	Allows the connection and level control of external powered speakers.
Talk to External	Mono	Allows board operator mic to talk to other devices, a logic command is associated
Talk to CR Audio	Mono	Allows the Talk to CR audio mix to drive an external destination

Monitor-related Inputs	Type	Comments
External Preview Input	Stereo	Allows an external path into the preview speakers, a logic command is associated
External Monitor Input 1	Stereo	Allows an external source to be monitored by CR or studio selectors
External Monitor Input 2	Stereo	Allows an external source to be monitored by CR or studio selectors

Source Inputs	Type	Comments
Microphone Input *n*	Mono	Typical installations have 2 microphone sources per studio
Analog Line Input *n*	Stereo	
Digital Line Input *n*	Stereo	

Source-related Outputs	Type	Comments
Feed-to-Source A *n*	Mono	Mono mix-minus output feeds the left side of a stereo connection. “Talk to...” function enabled.
Feed-to-Source B *n*	Mono	Mono mix-minus output feeds the right side of a stereo connection. “Talk to...” function disabled.

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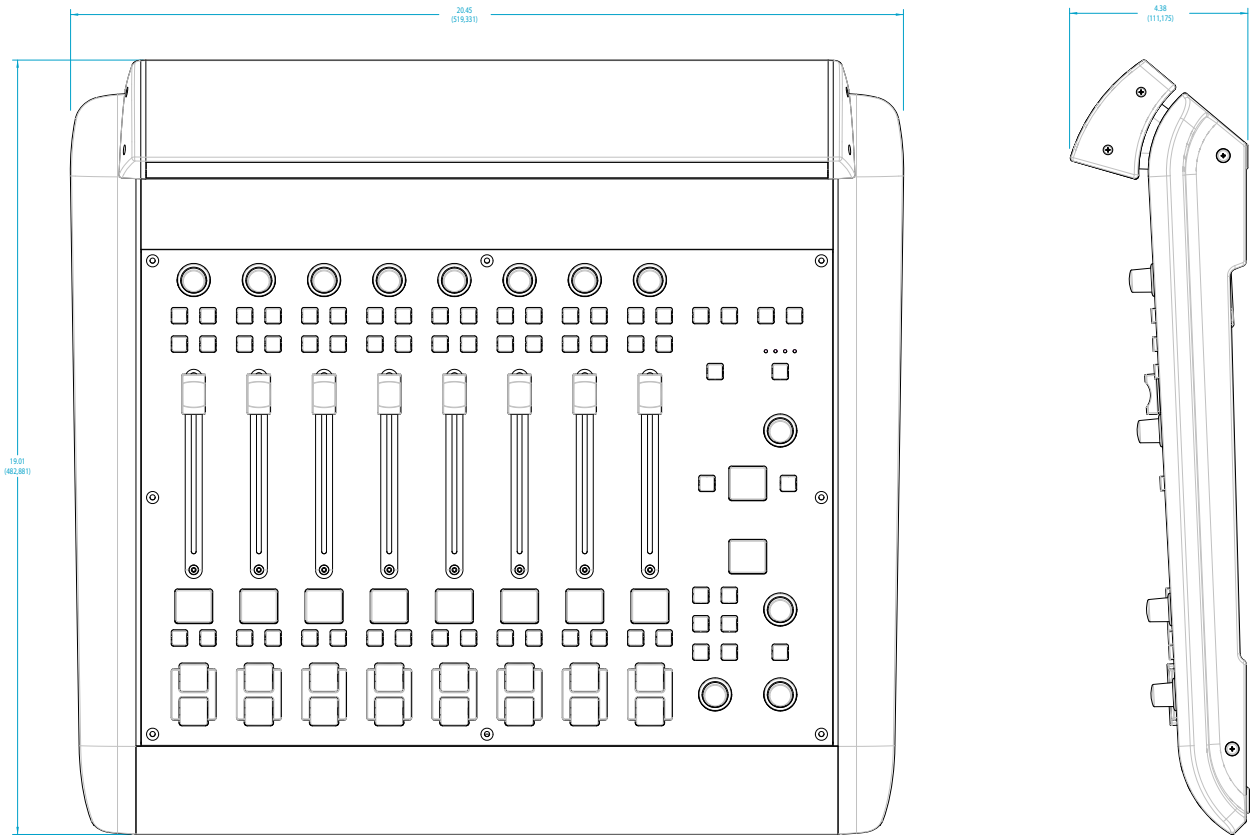
Radius Dimensions



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Marketers can't just

let it be — must fill spaces.

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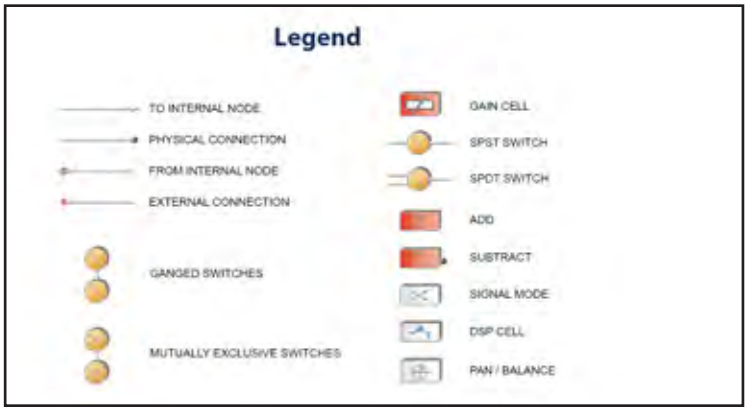
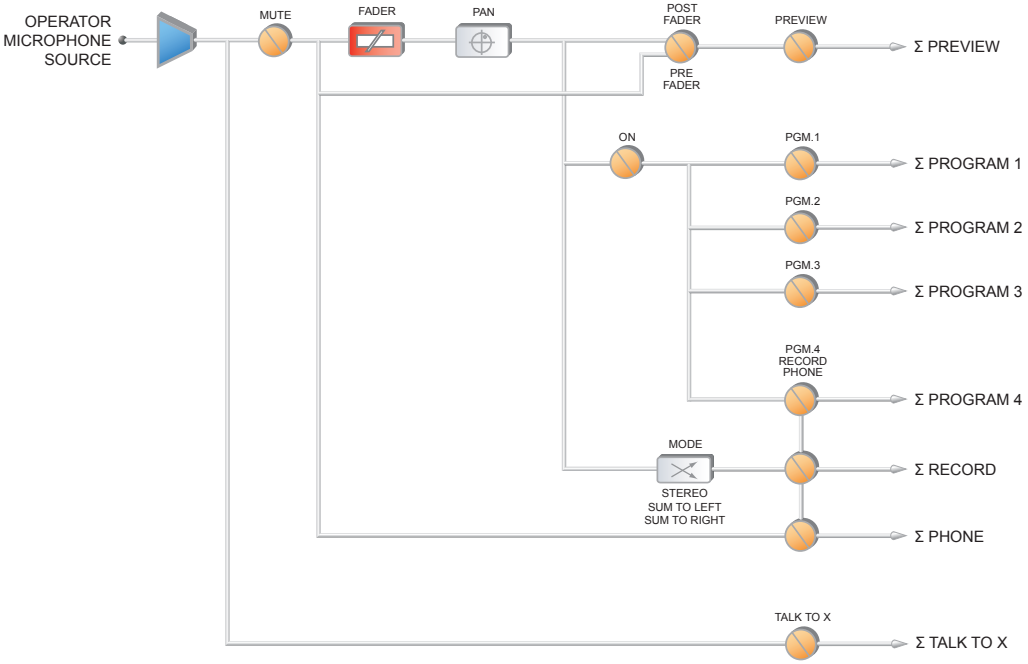
Appendix B: Radius Block Diagrams





Revised: January 2010

Figure B-1 Block Diagram - Operator Microphone

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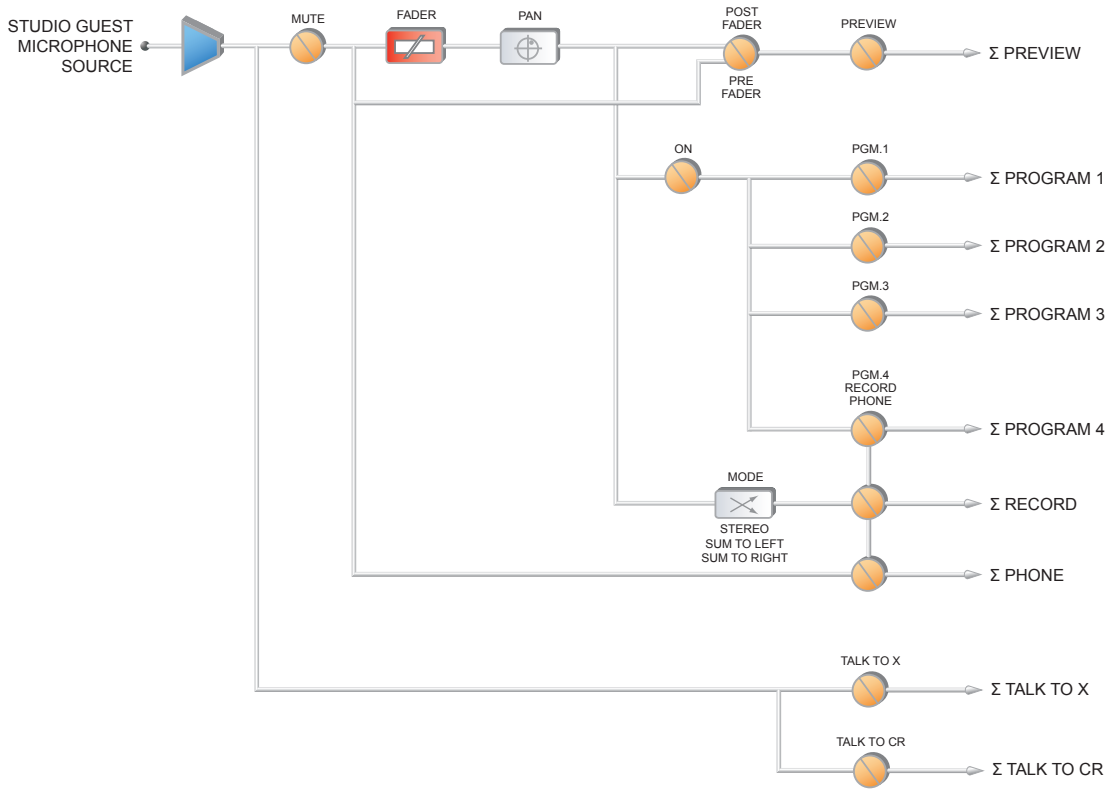
Radius Block Diagrams



Revised: January 2010

Figure B-2 Block Diagram - Guest Microphone

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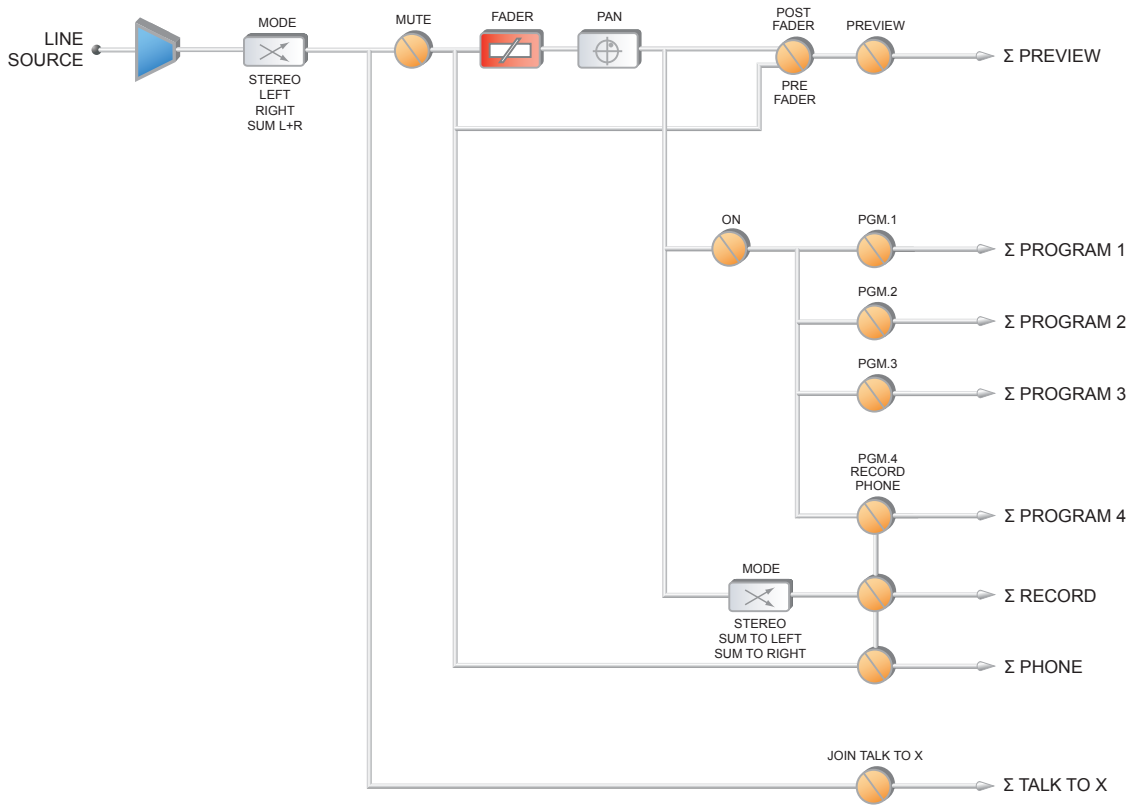
Radius Block Diagrams



Revised: January 2010

Figure B-3 Block Diagram - Line Source

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Radius Block Diagrams

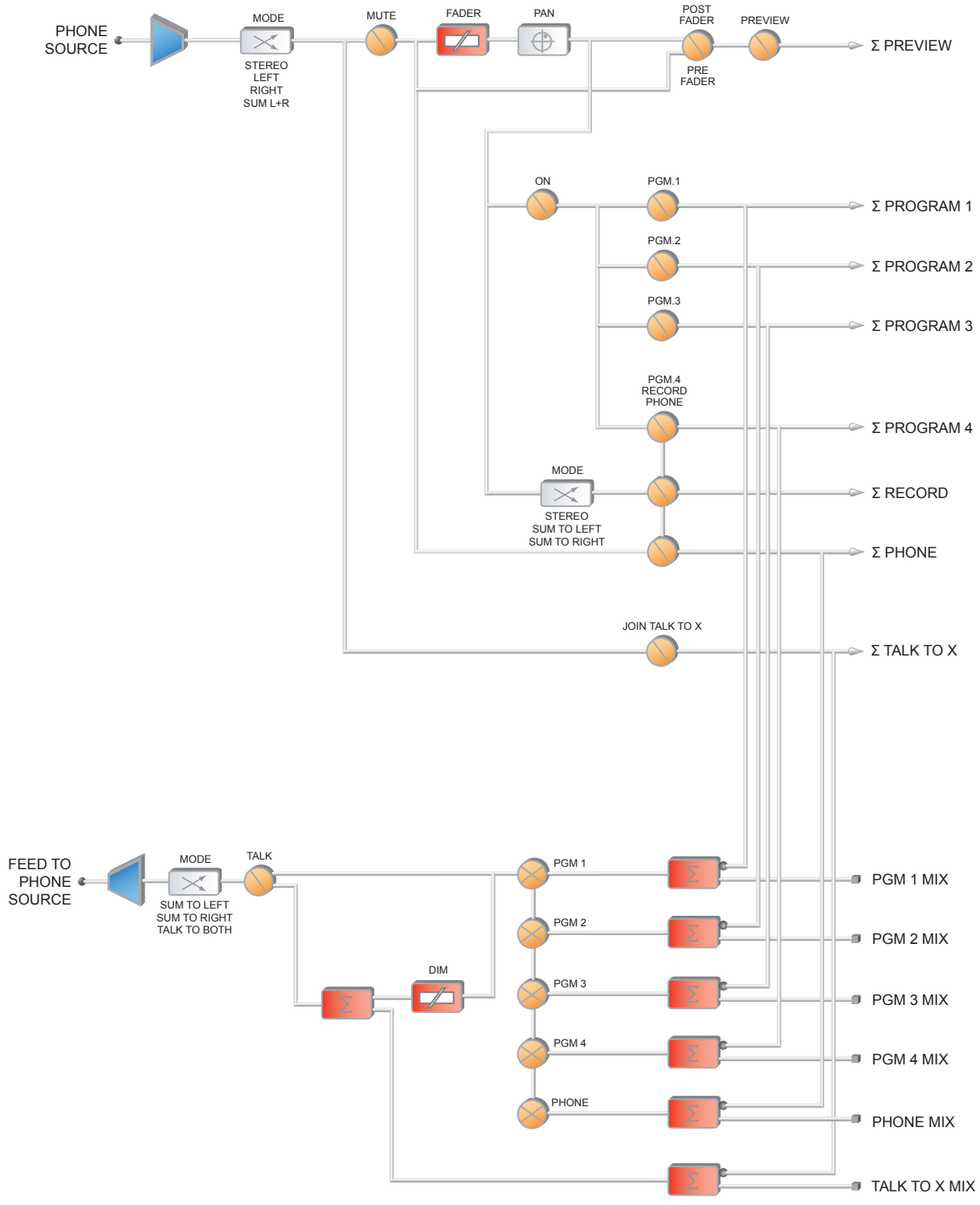

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Figure B-4 Block Diagram - Phone Source

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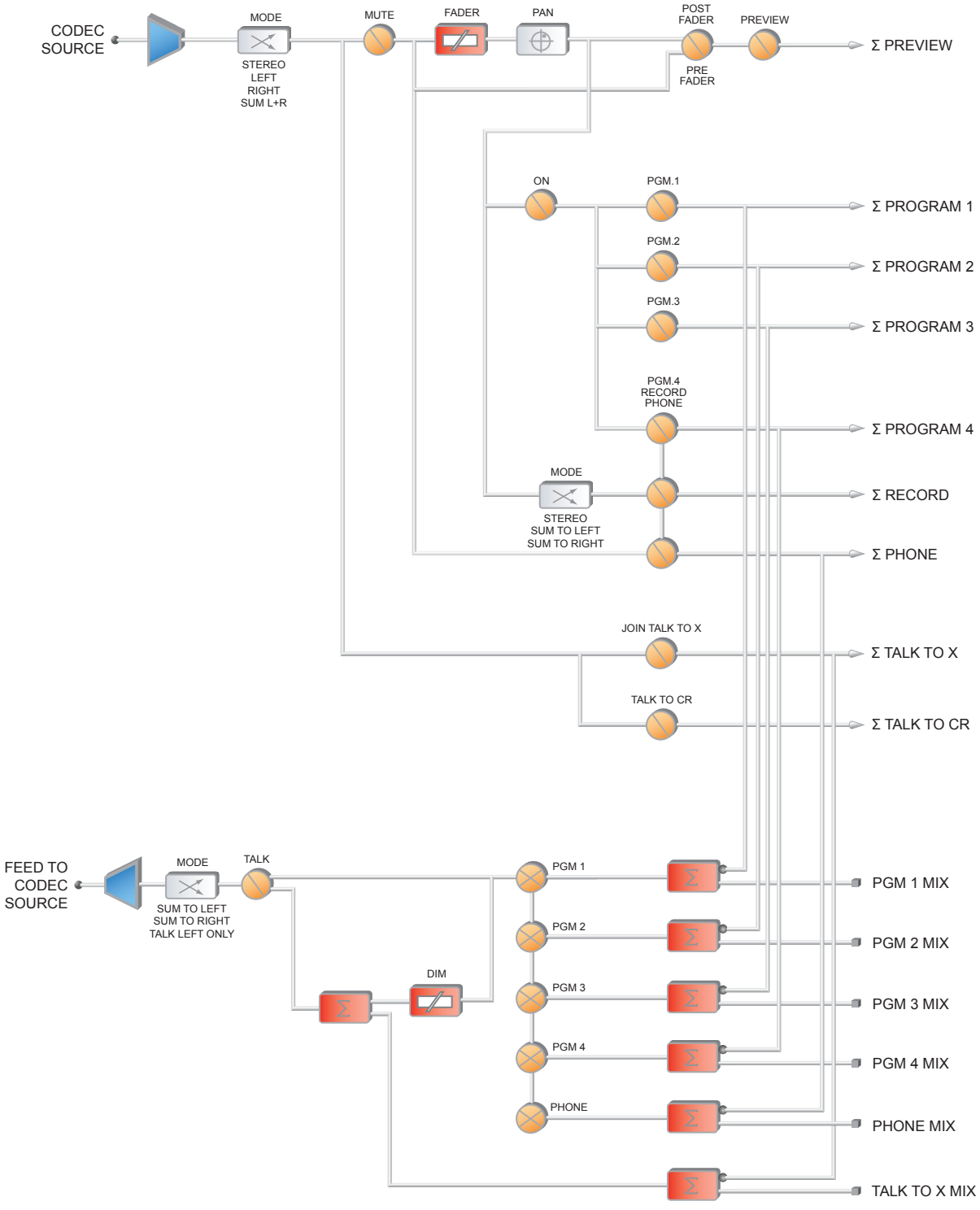
Radius Block Diagrams



Revised: January 2010

Figure B-5 Block Diagram - Codec Source

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Radius Block Diagrams

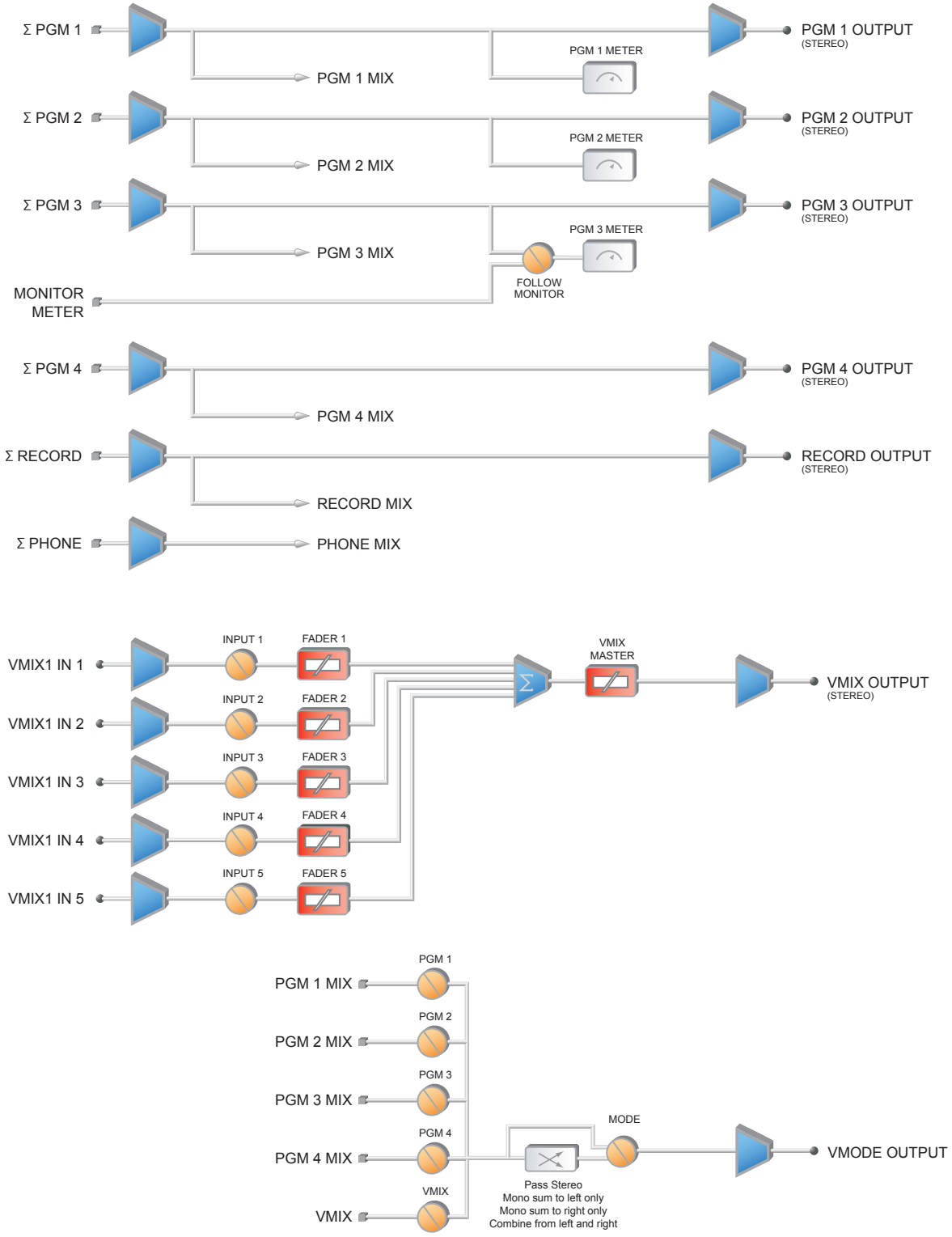

Figure B-6 Block Diagram - Master Output

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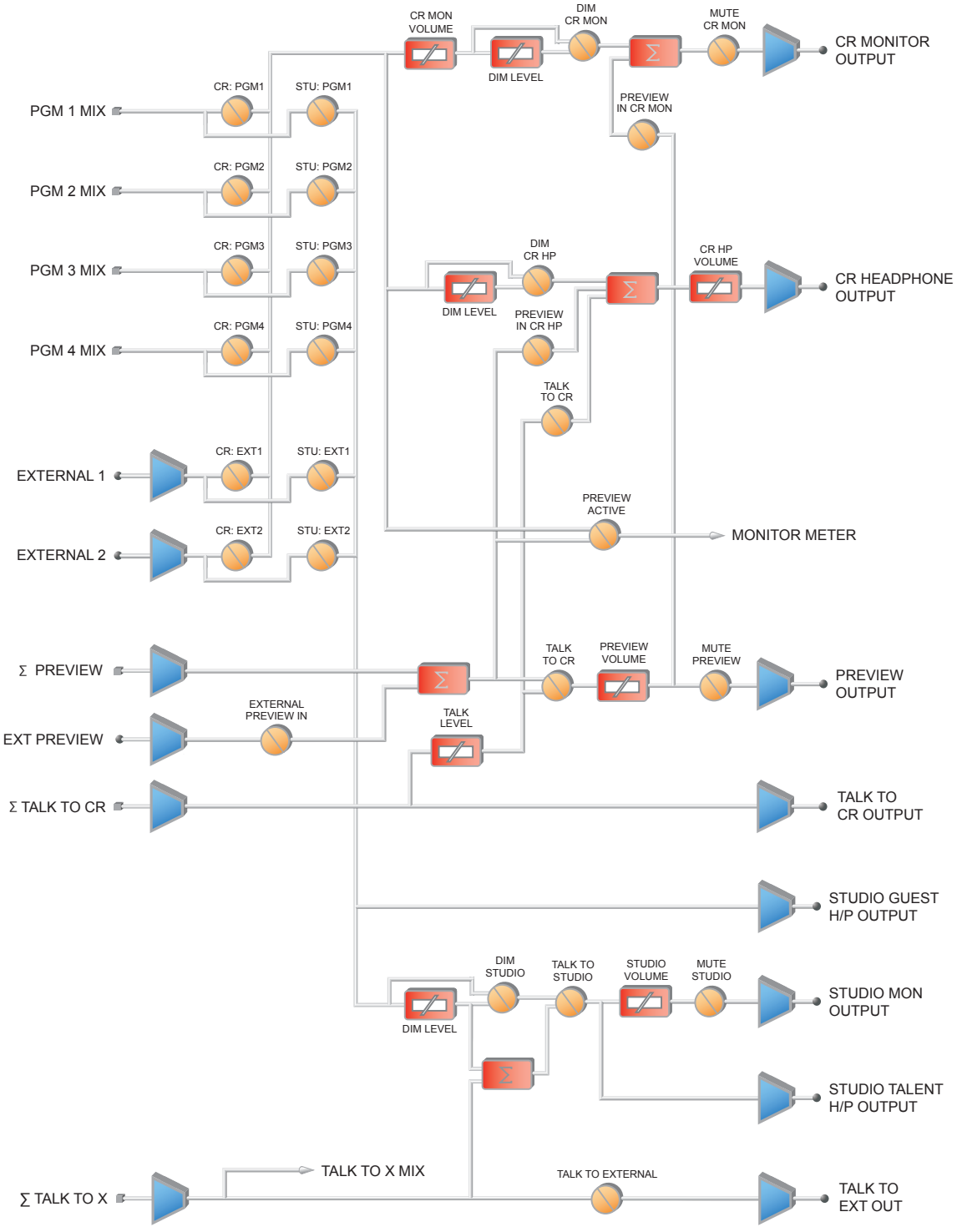
Radius Block Diagrams



Revised: January 2010

Figure B-7 Block Diagram - Monitor Output

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The cup steams, sweet scent

wafting lightly in the air.

Mmmmm... it is Earl Grey.

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Appendix C: Channel / IP Worksheets

The scalable nature of Axia Livewire audio networks makes it possible to construct systems of any size - from a single room to an entire multi-studio facility.

Since Livewire components – Audio Nodes, Elements, StudioEngines, PowerStation, Radius, etc. – are connected with Ethernet, each one requires a unique IP (Internet Protocol) address. IP addresses are four bytes long and are written in "dotted decimal" form, with each byte represented decimally and separated by a period. For example, in the IP address 193.32.216.9, the 193 is the value for the first byte, 32 for the second, etc. Since a byte can hold values from 0 to 255, this is the range for each decimal value. If you run a public network, Host IP addresses are assigned to your organization by your internet service provider and parceled out to individual host computers by your network administrator. He may give you this number to be entered manually, or could opt for DHCP (Dynamic Host Configuration Protocol) to let your computer get the address automatically from a pool. Because Livewire devices are permanently attached and because it is more desirable to know the IP address attached to a particular node (and perhaps assign them in some kind of logical pattern), we do not support DHCP for our hardware nodes. Therefore, you will need to enter an IP address into each node.

Livewire systems can support over 10,000 channels of audio; this necessitates giving each audio source or destination its own unique identifier, called *Channel ID numbers*.

As you can imagine, keeping track of so many unique IP addresses and Channel ID numbers is a big job. Therefore, on the next few pages, we've provided you with worksheets that you can use to keep track of the numbers you've assigned.

A logical convention that is often followed will link the Livewire channel number to the last octet of the device IP address. For example, if a node has an IP address of 192.168.1.12, the channel numbers assigned to the 8 sources of that node might be 1201 through 1208.

For a more detailed explanation of IP addresses, subnet masks and related issues, we highly suggest that you refer to "Network Engineering For Audio Engineers", Chapter 8 of our *Introduction to Livewire: IP Audio System Design Reference and Primer.*

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Axia IP Address Assignment Worksheet

IP Address	*Device Name*	*Location*	*Notes*
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Axia Audio Channel Assignment Worksheet - SOURCES

Channel ID Number	*Device Name*	*Device Location*	*Notes*

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Axia Audio Channel Assignment Worksheet - DESTINATIONS

Channel ID Number	*Device Name*	*Device Location*	*Notes*

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Warranty

Axia Audio Limited Warranty

This Warranty covers “the Products,” which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp., d/b/a Axia Audio (hereinafter “Axia Audio”).

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of five (5) years from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, AXIA AUDIO MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Axia Audio, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Axia Audio within the above-stated warranty period and warranty coverage must be authorized by Axia Audio. If Axia Audio authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Axia Audio, 1241 Superior Avenue, Cleveland, Ohio 44114.

Axia Audio at its option will either repair or replace the Product and such action shall be the full extent of Axia Audio’s obligation under this Warranty. After the Product is repaired or replaced, Axia Audio will return it to the party that sent the Product and Axia Audio will pay for the cost of shipping.

Axia Audio’s authorized dealers are not authorized to assume for Axia Audio any additional obligations or liabilities in connection with the dealers’ sale of the Products.

Axia Audio’s products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

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